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SayHi™

ES620 IP Phone

User Manual

Escene Communication Technology Co.Ltd

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1. Getting Started

About

SayHi ES620PE IP Phone series is a new generation intellectual desktop terminal. It brings in you a completely new and more efficient work experience in modern offices.

SayHi IP Phone series is a fashionable, beautifully shaped and high-tech product. Its excellent speech quality gives you an unprecedentedly comfortable talking and its rich function keys make the operation very convenient. One key dial can be easily achieved with proper settings. 4.3" high resolution LCD displayer provides you a friendly user interface. Benefit from which, more detail information can be displayed. With the help of the integrated browser, users can access any XML based information, and extend more commercials.

SayHi ES620PE IP Phone series support multi account registering and multi call management. Call Waiting, Call Queuing and Call Forward can be well implemented; meanwhile, it makes conference to be created very quickly. No matter how busy the calls will be, SayHi IP Phone can easily satisfy the requirement.

SayHi ES620PE IP Phone series supports 6 extend modules that totally contain 192 programming buttons. With the Busy Lamp Field function, operators can manage and monitor the calls more efficiently. It provides headset jack that compatible with professional microphone headsets, which will finely satisfy the call centre.

Features

- 4.3"480x272 Pixels HD LCD
- Customizable Screen
- HD Voice: HD Codec
- Advanced Call capability:
  8 VoIP Line/programmable buttons (dual-colors LED) and support BLF in programmable buttons.
- Expansion Module:
  Up to 6 EMS32 expansion modules and 192 programmable buttons, BLF supported
- Extensible Markup Language(XML)
- Support PoE and AC power adapter

Technical Parameters

Items	ES620

Screen	High-definition Color LCD with background light
	480x272 pixels (4.3 inch)
Line Buttons	8 VoIP Line buttons(dual-colors LED)
Function Buttons	4 Soft Keys, 4 Navigation buttons, OK button, C button Volume adjust, Hand-free, Mute, Headset, Message, Service, Directory, Menu, Conference, Transfer, Redial, Hold
VoIP Protocol	SIP 2.0
Network Protocols	HTTP, BOOTP, TFTP,*IEEE 802.1Q,*IEEE 802.1X
Codec	G.723.1 (5.3K 6.4K b/s), G.729 A(8Kb/s), G.711 A/U (64Kb/s), G722
QoS	TOS, Jiffer Buffer, VAD, CNG, G.168 (32ms)
Network	2*RJ45 10/100M Ethernet interfaces(LAN/PC) IP Assignment: Static IP or DHCP VPN(L2TP) DNS Clients (Primary and Secondary)
Conversation Mode	Handset, Headset or Hand-free Mode, Call center headset and 3.5mm headset supported 9-levels volume adjustment
Call Processing	Line status indicator (dual-colors LED), Multiple Accounts, Call Waiting, Call Queuing, Line Switching, Auto-answer Call Transfer, Call Holding, Call Pickup, One Button Dial, Redial Speed dial via Phone directory, Dial directly via Call record 3-way conference, DnD, Voice Mail, BLF (dual-colors LED)
Extension Interface	EXT: ESM32 programmable button module, support up to 6 ESM32s SOR: FXS, FXO USB(reserved)
Application	Public phone directory Private phone directory Other customizable applications XML
Security	Password Login Web Signaling encryption Media encryption

Management	Upgrade: HTTP/TFTP/Auto-provision Configurations: Phone/Web/Auto-provision Debug: Telnet/Phone/Web
Power Supply	Power adapter: AC 100-240V input and DC 12V/1A output PoE (IEEE 802.af)
Physical Features	Storage Temperature: 0℃-60℃ Operating Humidity: 10%-90% Size: 254mm*205mm*87mm Net weight: 1.37kg
Certifications	CE, RoHS

2. Connecting Your Phone

Your system administrator will likely connect your new SayHi ES620 IP Phone to the corporate IP telephony network. If that is not the case, refer to the graphic and table below to connect your phone.

1) Open the box ES620 IP Phone; carefully check the packing list, Packing List as follows:

Item	Counts
IP Phone	1
Handset	1
Handset Cord	1
Power adapter	1
RJ45 cable	1
CD	1
Quick Reference	1
Product certification	1

2) As shown in figure 2.1 and figure 2.2, Please plug Handset Cord into RJ11 interface(IP Phone and Handset), RJ45 cable into the LAN interface；IP Phone will automatically start if IP Phone with POE function.

3) The phone must work together with power adapter without POE support.

4) Connect your computer to PC interface of the phone with cable.

* More detailed description please refers to the *3.Phone overview-Understanding phone buttons and hardware*.

Figure 2.1 Interfaces of SayHi ES620

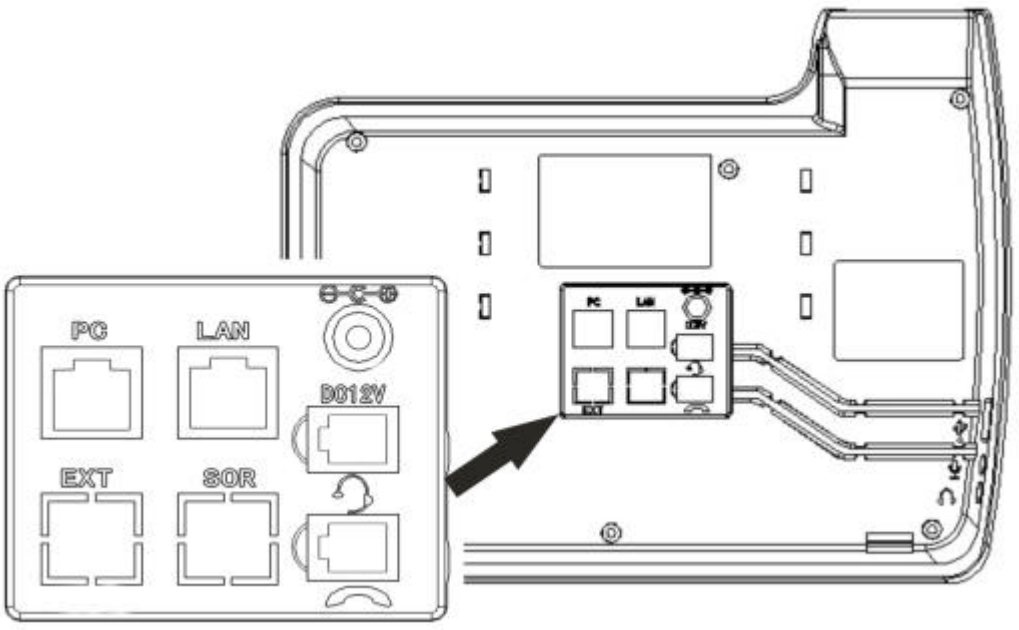
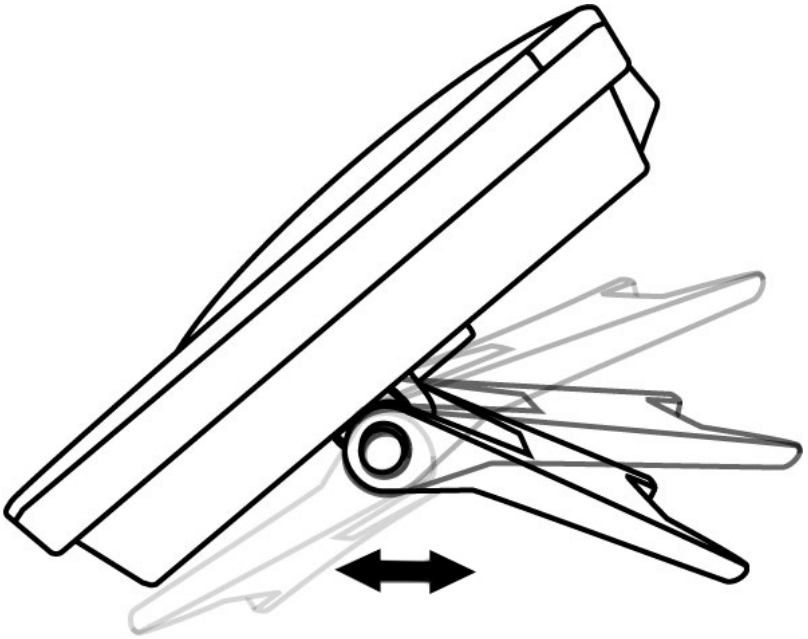


Figure 2.2 Footstand of ES620



5) If you want to connect an ESM (Expansion module), you could connect it (any interface on the ESM) to the EXT interface on the phone with a RJ-45 cable (a straight-through cable), and the second ESM can be connected to the first one too. It supports 6 ESMs.

3. Phone overview

Understanding Buttons and Hardware

From figure 3.1 to figure 3.2,you can understand buttons and hardware about SayHi ES620。

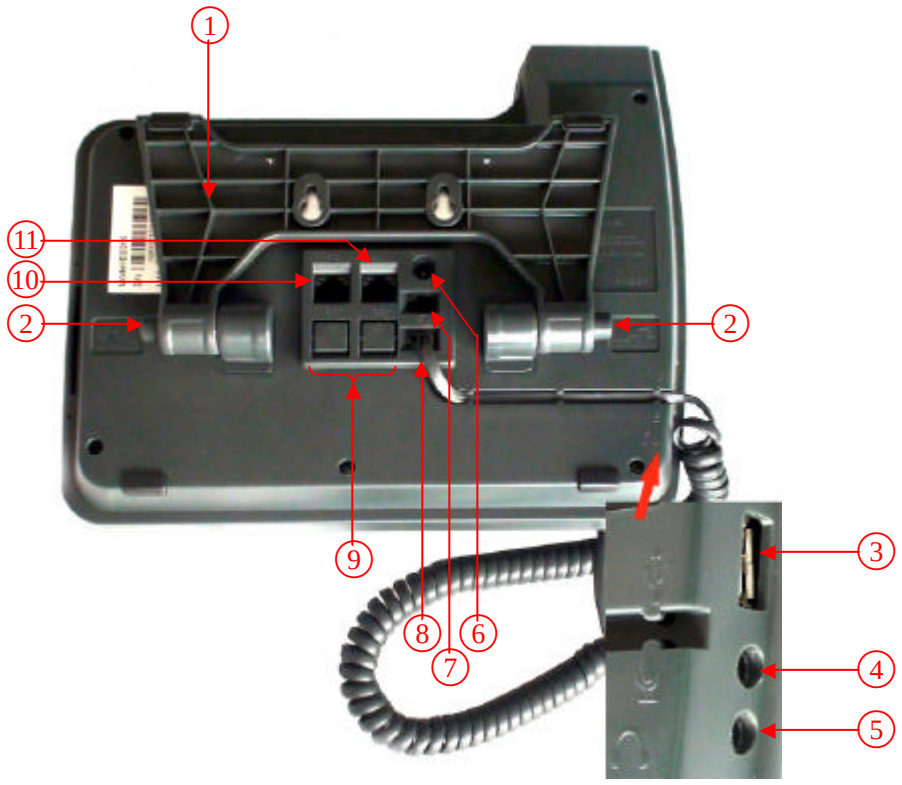
Figure 3.1 SayHi ES620 overview

Num	Buttons	Description
1		Headset button: Toggles the headset on or off Red means the feature is enabled.
2		Mute button: Toggles the Mute feature on or off. Red means the feature is enabled.
3		Message button: Typically auto-dials your voice message service. Red means have unread voice mail.
4	SERVICE	Service button: Open or Close the Services menu.
5	DIRECTORIES	Directories button: Use it to access call logs and corporate

		directories.
6	MENU	Menu button: Allows you to scroll through menus.
7		Volume button: Controls the volume and other settings.
8	CONFERENCE	Conference button: Connect calling / called party to the conference
9	REDIAL	Redial button: To Redial the last number.
10	TRANSFER	Transfer button: Transfer redirects a connected
11	HOLD	Hold button: Put a call on hold
12	0-9，*，#	Basic Call Handling: Press “#” send out a call(default)
13	Speaker button	Speaker button: Toggles the speakerphone on or off. Red, steady: Pick up and enter normal call.
14	Softkey	Each displays a softkey function, To activate a softkey, press the softkey button.
15	Line/Programmable Buttons	Line button: Select the phone line(Call or Answer) Different colors for different status: 1) Red, flashing: There is an incoming call. 2) Red, steady: Pick up and enter normal call. 3) Yellow-green, flashing: Holding call. 4) Yellow-green, steady: Active call. Programmable Keys: Hotline number can be used to bind in order to achieve speed dial; Turn on BLF: 1) Red, steady: Remote line is busying. 2) Yellow-green, steady: Remote line is idle.
16	C	Back button: Return to the standby interface;
17	Navigation button	“Up”: Open “Speed Call” list; “Down”: Open “Missed Calls” list:

		“Left”: Open “Received Calls” list; “Right”: Open “Dialed Numbers” list
18	OK	OK button: To confirm the action;
19	Hands-free speakerphone	Hands-free voice of the output.
20	LCD screen	480*272 pixel Color high-definition display.
21	Status light	Red flashing: There are incoming call ; Red, steady: Missed Calls, or phone busy;
22	Hands-free microphone	Hands-free voice of the output.
23	Line Expansion Module	Maximum support 6 ESM32 line expansion module (the total of 192 programmable buttons)

Figure 3.2 Interfaces of SayHi ES620



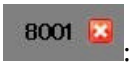
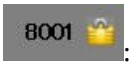
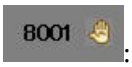
Num	Hardware	functions
1	Footstand	Hold up phone
2	Footstand button	Press buttons at the same time to adjust the angle
3	Reserved for USB port	Enhanced scalability
4	Microphone port	Connect the Microphone
5	Headphone port	Connect the Headphone
6	Power port	12VDC
7	Headset port	Support RJ11 interface connection
8	Handset port	Connect the Handset
9	EXT port	EXT: ESM interface;
10	LAN port	Connect to a LAN interconnecting device
11	PC port	Connect to a local PC

Understanding Phone Screen Features

This is what your main phone screen might look like with an active call:

Figure 3.3 LCD of SayHi ES620

Num	Screen	Functions
1	Time and Data	Display current time and data
2	Service status	1) : Headset turned on 2) : There are Missed calls 3) : Auto-answer turned on 4) : Mute turned on 5) : DND turned on
3	Network connection status	Phone LAN interface network connection status display: 1) : Network connection is successful 2) : Network is disconnected.
4	Menu Functions Page	1) : Phone Standby/Calling page 2) : Main menu/Call history page

		3) : Phone Book page 4) : System Services page
5	Primary phone Line	Displays the phone number(extension number)for your primary Phone line.
6	Call activity Area	Displays calls per line, including caller ID, for the highlighted line.
7	Missed calls tips	Show the number of missed calls.
8	Line status/ Speed Call	1) : Line is successfully registered to a SIP server 2)  : Line is not successfully registered to a SIP server 3)  : DND turned on in this line 4)  : Line is held 5) : Peer-to-Peer status 6) : dialing 7) : talking 8) : ringing 9) : Transferring a call 10) : 3-ways conference 11) : speed dial key
9	Softkey labels	Each displays a softkey function. To activate a softkey, press the softkey button.

4. Basic Call Handing

You can perform basic call-handling tasks using a range of features and services. Feature availability can vary; see your system administrator for more information.

Note: The bold type of the following text and following a "button" in table signifies the phone's button (for example, **OK** button), and the NewCall signifies softkey.

Placing a Call

Here are some easy ways to place a call on SayHi ES620 IP Phone: :

<table>
<tr><th>If you want to...</th><th colspan="2">Then...</th></tr>
<tr><td>Place a call using the handset</td><td>Pick up the handset;</td><td rowspan="3">--1）Hear the dial tone；
--2）The first line light and display 8001 ；
--3）Enter number；
--4）Press "#" button(default);
or press Send;
or wait five seconds(default)
Then send the call;</td></tr>
<tr><td>Place a call using a Speakerphone</td><td>Press Speaker button;
or Programmable buttons;
or NewCall;</td></tr>
<tr><td>Place a call using a headset</td><td>Press Headset button.</td></tr>
<tr><td>Redial</td><td colspan="2">--Press REDIAL button to dial the last number
-or press Navigation button-Right > "Dialed number", select a number, and press Dial or OK button.</td></tr>
<tr><td>Dial from the Directory on your phone</td><td colspan="2">--1) Press MENU or OK button > "Call history", you can select "Missed calls", "Received calls" and "Dialed numbers",
- or press Navigation button (in Standby interface) > select "Missed calls" (down), "Received calls" (left) and "Dialed numbers" (right))；
--2) Then press OK button or Dial.</td></tr>
<tr><td>Place a call while</td><td colspan="2">--1) Press Hold button or Hold;</td></tr>
</table>

Another call is active	--2) Enter a number; --3) Press '#' button (default) ; -or press Send to send the number.

Tips

• You can dial on-hook, without a dial tone (pre-dial). To pre-dial, enter a number, and then go off-hook by lifting the handset or pressing Send, **Headset** or **Speaker** button.

• If you make a mistake while dialing, press **C** button to erase digits.

Answering a call

You can answer a call by simply lifting the handset，or you can use other options if they are available on SayHi ES620:

If you want to...	**Then...**	
Answer with a handset	--1) Your phone ring; --2) **Line** button of the ringing line is Red and flashing, Light strip is Red and flashing; --3) Status of the line on the phone is 8001.	--Pick up the handset
Answer with the speakerphone (Non-headset mode)		--Press **Speaker** button -or press the **Line** button flashing Red, -or press Answer
Answer with the a headset		--Put on headset, press **Headset** button so that the status light is Red, and then do as using speakerphone
Switch from a connected Call to answer a ringing call	--1) Another **Line** button is Red and flashing, Light strip is Red and flashing; --2) Press the flashing **Line** button to answer (at this time, the original call will be hold.)	
Auto-answer	--1) Press **MENU** or **OK** button > "Function setting" > "Auto answer"; --2) Select "Enable"; --3) Your phone answers incoming calls automatically after a few rings.	

Ending a Call

To end a call, simply hang up, here are some more details:

If you want to...	Then...
Hang up while using the Handset	--Return the handset to its cradle, -or press
Hang up while using the Speakerphone	--Press **Speaker** button that is Red , -or press **Line** button for the appropriate line, -or press
Hang up while using the Headset	--Press **Handset** button, （Do not keep the headset mode) , -or press (keep the headset mode)
Hang up one call, but preserve another call on the other line	--Press , -or refer to the above three methods

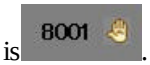
Using Hold and Resume

You can hold and resume calls. You can take a call in one line at anytime, and the other lines would be hold. As a result of that, you can switch different calling line on our phone.

If you want to...	Then...
Put a call on hold	--Press **HOLD** button, -or press
Hold a line and switch to another line	Press another **Line** button for the appropriate line
Resume a call on current line	--Press appropriate line button,
Release a call on different line	Select the line want to release hold, press the line, so recovery;

Tips

• Engaging the Hold feature typically generates music or a beeping tone.

• A held call is indicated by the Yellow-green and flashing Line button. And line status is 

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Transferring Calls

Transfer redirects a connected call. The target is the number to which you want to transfer the call。

If you want to...	Then...
Talk to the transfer recipient before transferring a call (consult transfer)	--1) Press **TRANSFER** button or press Transfer; --2) Enter number; --3) press "**#**"（default) , -or press Send then transfer the call, -or wait five seconds(default)then transfer the call
Transferred to idle lines or other numbers without talking to the transfer recipient (Blind transfer)	--1) Press **TRANSFER** button or Transfer ; --2) Press Blind; --3) Enter number; --4) Press "**#**"（default) -or press Send, then transfer the call; -or wait five seconds(default)then transfer the call
Blind transfer to the held line	--1) Press **TRANSFER** button or press Transfer; --2) Press the **Line** button of held line

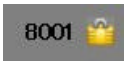
Using Mute

With Mute enabled, you can hear other parties on a call but they cannot hear you. You can use mute in conjunction with the handset, speakerphone, or a headset。

If you want to...	Then...
Toggle Mute on	Press **Mute** button, then the button is Red
Toggle Mute off	Press **Mute** button, then the button light off

Do Not Disturb

You can use the Do Not Disturb(DND)feature to block incoming calls on your phone with a busy tone(Can also be set to their voice mail or other extension numbers, etc.)。

If you want to...	Then...
Enable global DND	--1) Press ; --2) All enabled line on the phone would changes to  status.
Enable DND on a single line	Press **MENU** or **OK** button > "Function setting" > "DND" > (select line) "Enable"
Disable DND	--Global DND enabled, press to disable global DND; --Line DND enabled, press twice , -or press **MENU** or **OK** button > "Function setting" > "DND" >(select line) "Disable"

3-way Conference

You can establish a three-party conference, during the conversation three phone parties can communicate with each other.

If you want to...	Then...
Invite the transfer recipient into a conference in a transferring	--1) When the transfer recipient answer the call, press **CONFERCENCE** button or on your phone; --2) Then the held one, transfer recipient and you will be into a conference.

Invite the third party into a conference in a active call	--1) Press **CONFERENCE** button or in an active call; --2) Enter the third party number **;** --3) After connected the third party, press **CONFERENCE** button or again
establish a conference with held line	--1) when one phone line is holding on and the other line is busy; --2) Press **CONFERENCE** button, -or Press Soft key --3) press the held line's programmable button, the 3-way Conference will establish.

5. Advanced Call Handling

Using the phone book

You can store a large number of contacts in your phone's directory. You can add, edit, delete, dial, or search for a contact in this directory.
The personal phonebook can storage 300 contact entry,and the enterprise phonebook can storage 800 contact entry.

If you want to…	Then…
Add Contacts	--1) Press PhoneBook, -or press **MENU** button > "Phone book">"Personal phone book", -or press **Directories** button > "Phone book">"Personal phone book"; --2) Press Modify; --3) Select "Add contact", press **OK** button or OK; --4) Use the navigation keys to select content, press **OK** button or Modify to set and modify: -Name: set the name of contact, -NO.1-3: you can set up 3 contacts' numbers, -Group: the contacts be divided into different user's groups --5) Press Submit soft key to complete
Add group	--1) Press PhoneBook soft key, -or press **MENU** button > "Phone book">"Personal phone book", -or press **Directories** button > "Phone book">"Personal phone book"; --2) Press Modify soft key; --3) Select the "add group" then press **OK** button or OK; --4) Use the navigation keys to select content, press **OK** button or

	Modify to set and modify: -Group name: name of the group --5) Press Submit soft key to complete
Modify group	--1) Press PhoneBook soft key, -or press **MENU** button > "Phone book">"Personal phone book", -or press **Directories** button > "Phone book">"Personal phone book"; --2) Press Modify soft key; --3) Select the "Modify group" then press **OK** button or press OK; --4) Select the group you want to modify, press the **OK** button or Modify to set and modify, press Submit to save the change
Delete group	--1) Press PhoneBook soft key, -or press **MENU** button > "Phone book">"Personal phone book", -or press **Directories** button > "Phone book">"Personal phone book"; --2) Press Modify soft key ; --3) Select the "Delete group" or **OK** button or OK ; --4) Select a group you want to delete, press **OK** button or OK
View/Edit Contacts	--1) Press PhoneBook soft key, -or press **MENU** button > "Phone book">"Personal phone book", -or press **Directories** button > "Phone book">"Personal phone book"; --2) Select "View ALL", -or select a contact who are belong to different group;

	--3) Select the contact, press the **OK** button or View (to edit the contact's information, press **OK** button or Modify)
Call from phone book	--1) Press PhoneBook soft key, -or press **MENU** button > "Phone book">"Personal phone book", -or press **Directories** button > "Phone book">"Personal phone book"; --2) Select "View ALL", -or select a contact who are belong to different group; --3) Select a contact, then press Dial, (If there are multiple numbers of one contact, press Dial to enter the interface of "call options", select the one you want to call and press Dial)

Using Call Logs

Your phone maintains records of your missed, placed, and received calls.

If you want to…	Then…
View your call logs	--1) Press **MENU** button > "Call history" > "Missed Calls", "Received Calls", or "Dialed numbers" --2) Use the navigation keys to view the call record information.
Dial from a call log	Please refer to the previous part *4.Basic call handing – Placing a call.*
Erase your call logs	--1) If you want to delete a call record, you have to select this record from the logs and press Delete; --2) If you want to delete an entire call record list, you have to select this record list from the logs and press DelALL

Tips

• Each call log store up to 100 entries on SayHi ES620 IP phone.

6. Keypad Instruction

SayHi series IP phones are can be configured in two ways. The first you can use the phone keypad where you can settings for you IP phones, the other you can log in to User Options web pages where you can settings for you IP phones.

Use phone keypad to setting. Press **MENU** or **OK** button to the main menu, Use the navigation keys to select menu, press **OK** button to confirm menu selections, press **C** button or

to delete input information.

Language

SayHi ES620 IP Phone supports Simplified Chinese and English.

If you want to…	Then…
To change the language via phone interface	--1) Choose “System setting” > “Phone setting” > “Language”; --2) Scroll through the list of available languages. --3) Press **OK** button or when the desired language is highlighted. The language appears on the graphic display will be changed to the one you chose. --4)Or press OK button or MENU,navigate to Language,press OK,then select the language you want,press OK it will be changed to the one you chose.

SIP Account Settings

SayHi ES620 series IP phone make calls based on sip accounts, SayHi ES620 series IP phones can support 8 independent SIP account, each account can be configured to different SIP server.

If you want to…	Then...
Create an sip account	--1) Choose “System settings” > “Advanced settings”； --2) Enter the password required（The default is empty）； --3) Choose “SIP” > “Accounts sip”； --4) Choose one of the account you want to setting, you can configure

	the following parameters -**Enable account**: choose Enable -**Account Mode:** VOIP -**Display Name**: The name displayed on the screen -**User Name**: the account matched with the SIP server. （extension number）， -**Authen usr**: the Authenticated users matched with the SIP server. （The default With the same account） -**user pwd**: the user password matched with the SIP server -**Description**: description of this account， -**SIP1**: the primary SIP server，By default all calls through the server， -**SIP2**: the secondary SIP ，When the primary server is unavailable ,use the SIP server -**Refresh time**: Registration refresh interval, the minimum value is 20 The default value is 3600. -**Con type**: the protocol use to communicate with SIP server. -**Amount of used lines**: the lines the extension used,can use 1-8,default is 2. --5）Set up the above parameters，Press Submit softkey to saves settings，Complete the account creation； * **Note**: the parameters with the * mark must be set.
Disable sip account	--1) Choose “System setting” > “Advanced setting”； --2) Enter the password required（The default is empty）； --3) Choose “SIP” > “Account sip”； --4) Choose “Enable account” > “Disable”； --5）Press Submit soft key

Network Setting

If you want to…	Then…
network setting	--1) Choose “System settings” > “Advanced settings”； --2) Enter the password required（The default is empty）； --3) Choose “Network”, you can configure the following parameters: -**Type**: static IP or DHCP or PPPOE -**IP**: enter IP address ，Note: Do not duplicate the ip address with other devices on the network -**Mask**: enter appropriate sub mask -**GW**: enter appropriate gateway - **DNS1**: enter IP address of the primary DNS server - **DNS2**: enter IP address of the secondary DNS server -**Web port**: the default Web port is 80,if you change it(for example change it to 88),you must use IP and Web port to login the web page (for example http://192.168.0.200:88).It will take effect on next reboot. -**Telnet port**: the default Telnet port is 23,if you change it(for example change it to 2003),you must use IP and Telnet port to login the manage page (for example telnet 192.168.0.200:2003).It will take effect on next reboot.

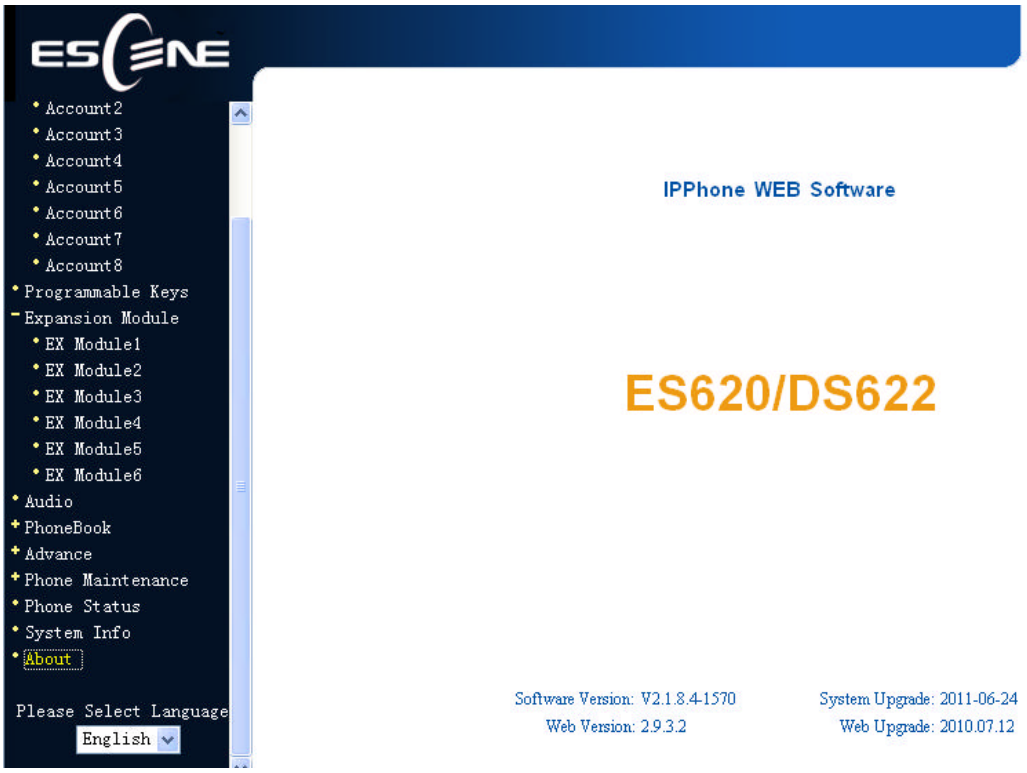
Customizing Rings and Volume

If you want to…	This…
Change the ring tone	--1) Choose “System settings” > “Phone settings” > “Ring type”； --2) Press navigation to choose ring tone； --3）Press softkey to choose a ring tone to play a sample of it.

	Press softkey to Stop Playing Press OK or softkey to set the ring tone, Press softkey to return to previous menu.
Adjust the volume level	--1) Choose “System settings” > “Phone settings” > “Volume settings” --2） You can adjust the volume level of following types -**Ring volume**: Phone call ring volume, -**Handset volume**: Handle output volume， -**Handset mic volume**: Handle input volume， -**Speaker volume**: Hands-free speaker output volume， -**Speaker mic volume**: Hands-free input volume， -**Headset volume**: Headphone output volume， -**Headset mic volume**: Headset microphone input volume

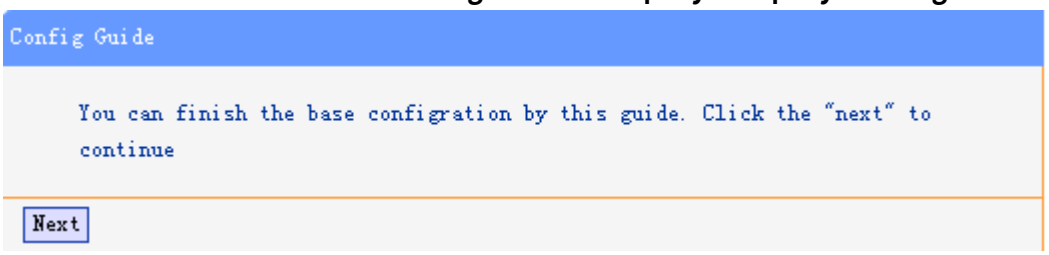
7. Web Settings

We can configure the IP Phone more handy through web setting. Press OK button on the keypad of the phone to enter the status page and find out the IP address of IP phone. Enter it (for example you can enter the default ip of the phone: http://192.168.0.200) into the address bar of web browser. The default login name and password are both "root".

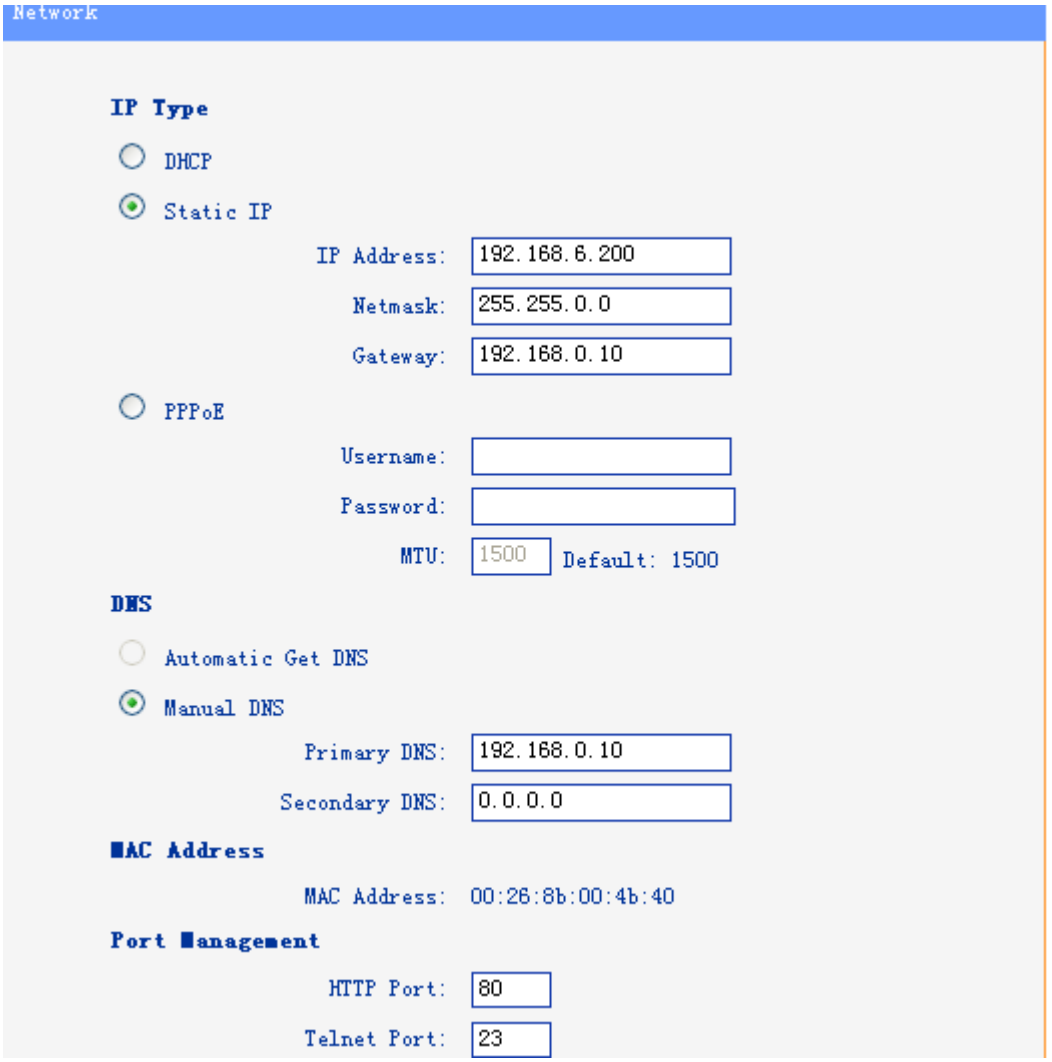


Config Guide

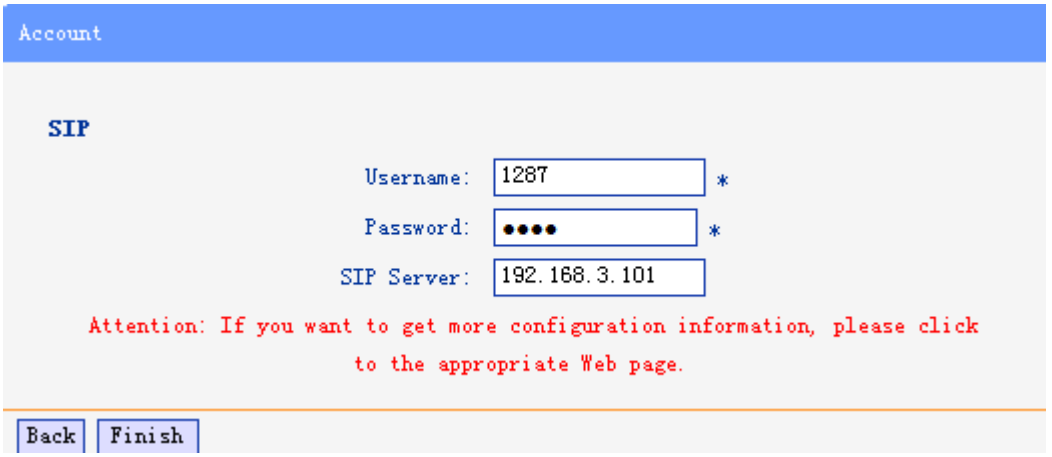
You can finish the base configration step-by-step by this guide.



When press 'next',you can configure the Network parameters for the phone,



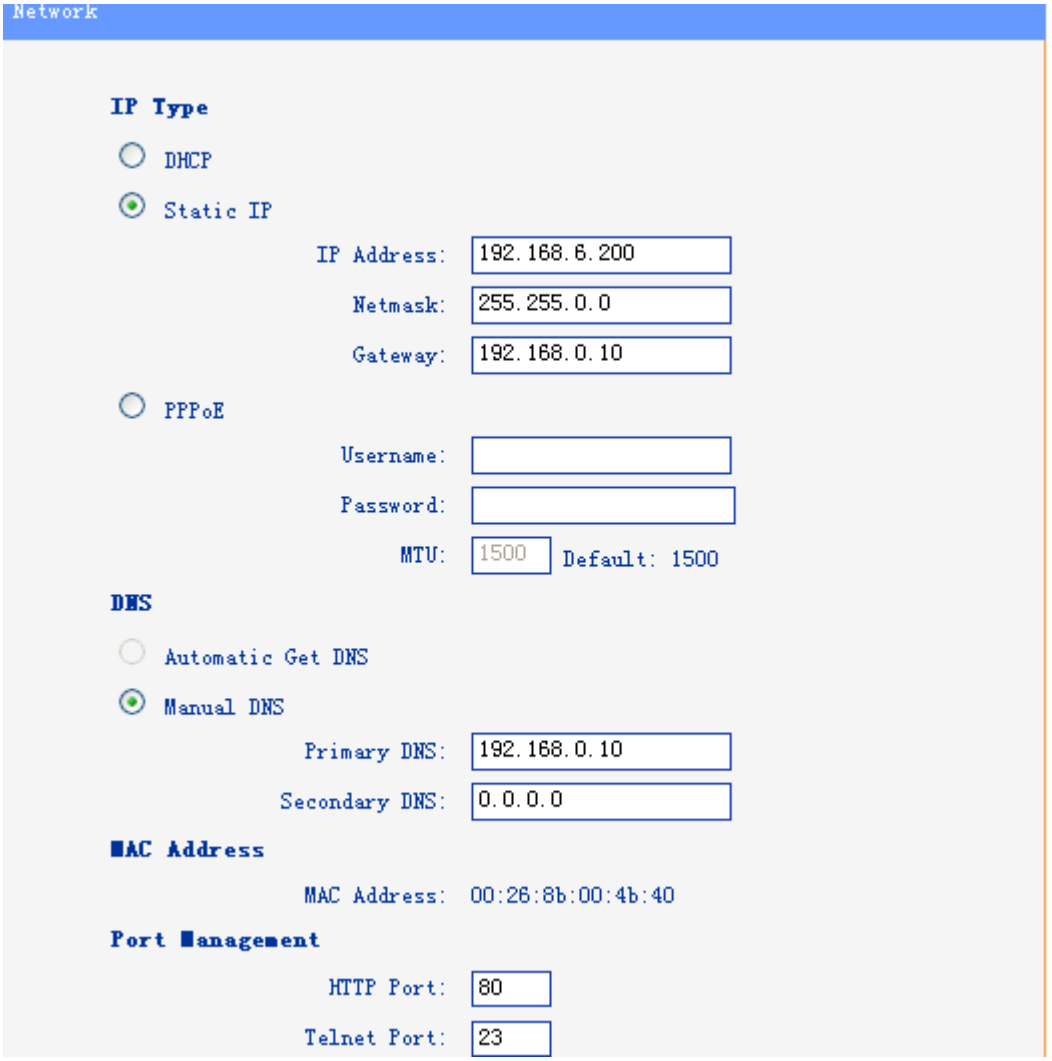
After config the network parameter,press next,then you can config sip account for the phone.



Press Finish,the base configuration of the phone is complete,now you can use the phone to call with sip.

Network

You can config the network parameters for the phone on the web page.



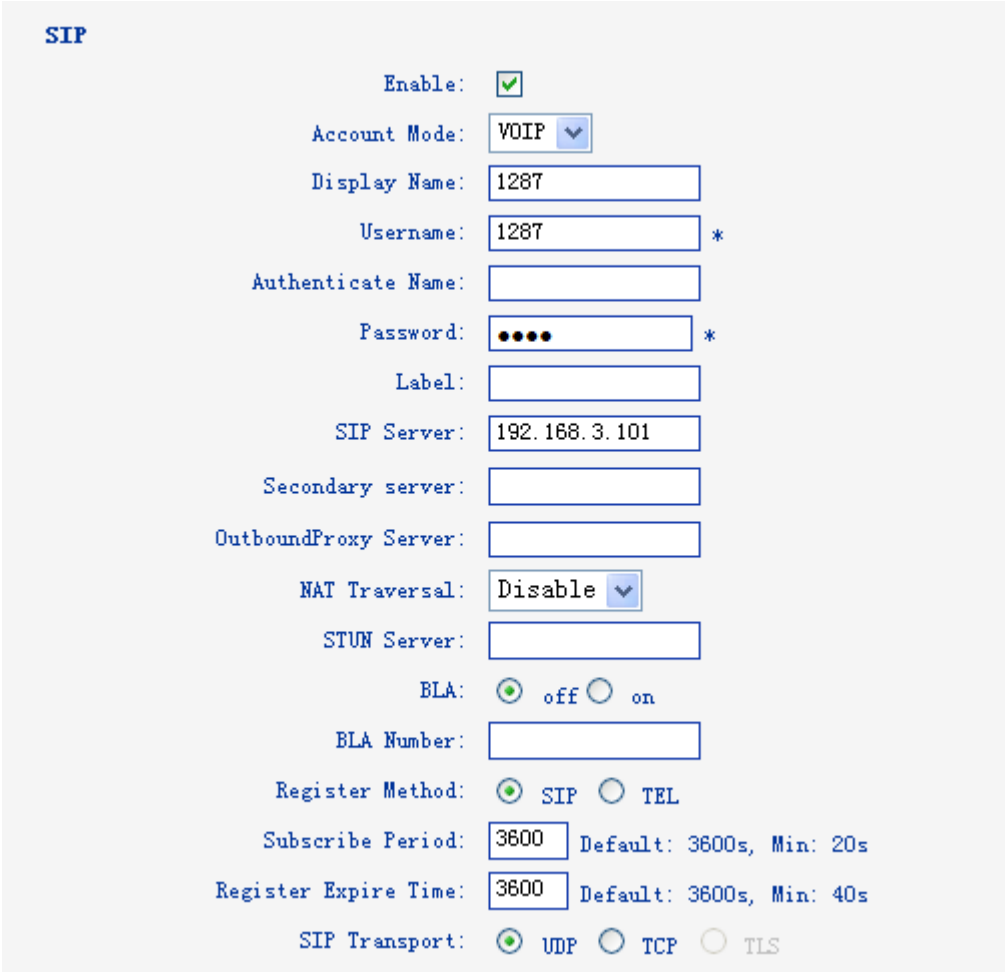
Choose network,you will find the following parameters:

Field	Description
DHCP	Config the phone get ip info from DHCP server
IP Address	Config the ip manual for phone
Netmask	Config the netmask manual for phone
Gateway	Config the gateway manual for phone
Username (pppoe)	The pppoe username
Password (pppoe)	The pppoe password
MTU (pppoe)	The mtu for pppoe,default is 1500
Primary DNS	The primary DNS server
Secondary	The secondary DNS server

MAC Address	Display the MAC of the phone
HTTP Port	The default web port is 80,if you change it(for example change it to88), You must use IP and Web port to login the web page(for example http://192.168.0.200:88). It will take effect on next reboot.
Telnet Port	the default Telnet port is 23,if you change it(for example change it to 2003),you must use IP and Telnet port to login the manage page (for example telnet 192.168.0.200:2003).It will take effect on next reboot.

SIP Account

The phone attempts to register to the SIP server using the account/registrar data provided by the automatic or manual initialization.



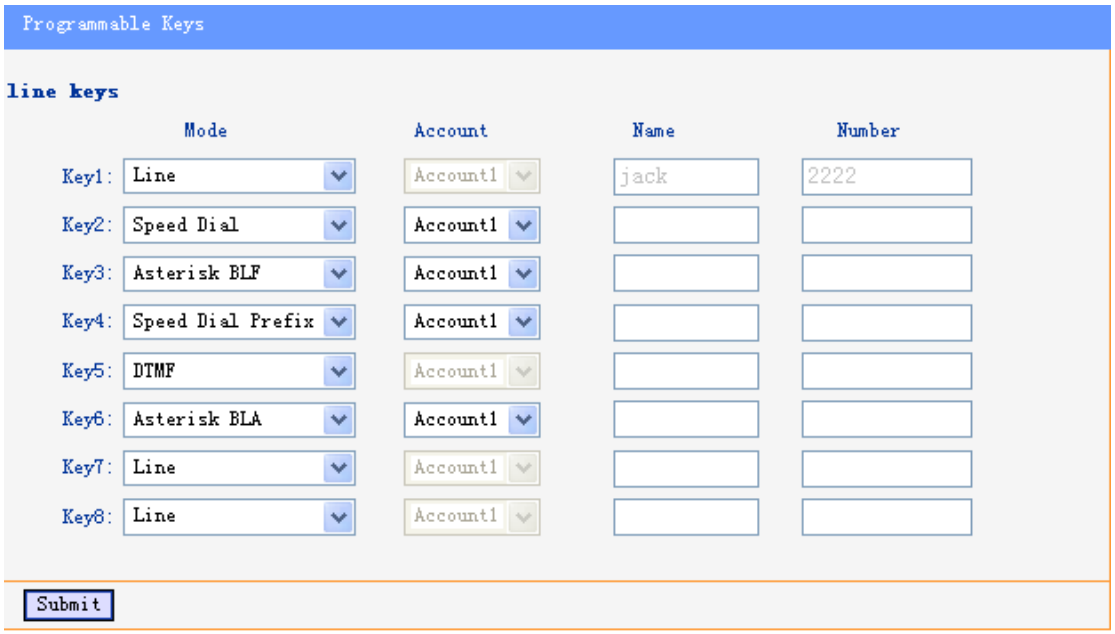
Choose one Account, you will find the following parameters:

Field	Description
Enable	You can choose on/off to enable/disable the line.
Account Mode	You can choose VOIP

Display Name	It is showed as Caller ID when making a phone call
Username	It is a username provide by SIP Server
Authenticate Name	It is authenticated ID for authentication
Password	It is a password provide by SIP Server
SIP Server	Server for registration, provided by administrator
Register Expire Time	IP phone automatically registered every time
Amount Of Line Account Used	The line key of account used,default is 2

Programmable Keys

In the Programmable Keys option,you can use all the 8 line keys of ES620 as programmable keys.



Choose Programmable Keys,you will find the following parameters:

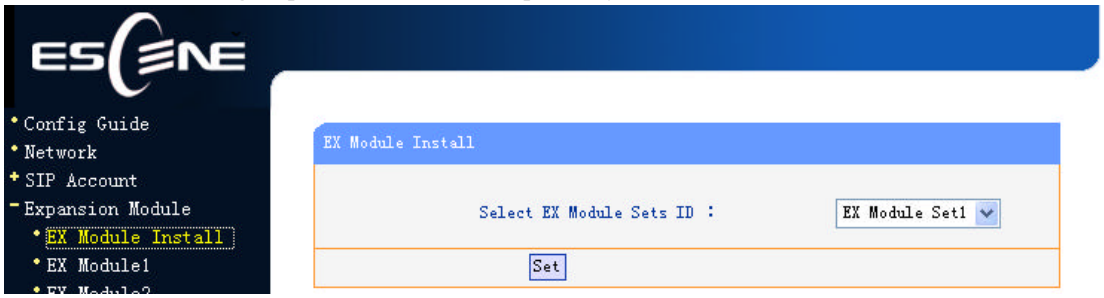
Field	Description
Line (Mode)	Use specific Key as Line key,default is Line key
Speed Dial(Mode)	Use specific Key as Speed Dial key
Asterisk BLF(Mode)	Use specific Key as BLF key
Speed Dial Prefix(Mode)	Use specific Key as Speed Dial Prefix key
DTMF	Use specific Key as DTMF key
Asterisk BLA	Use specific Key as BLA key
Account	Select a SIP account relating this programmable key,that is you can dial to the specific number from this SIP account

Name	The name for the programmable key
Number	The phone number relating this programmable key,that is the number you use the programmable key to dial

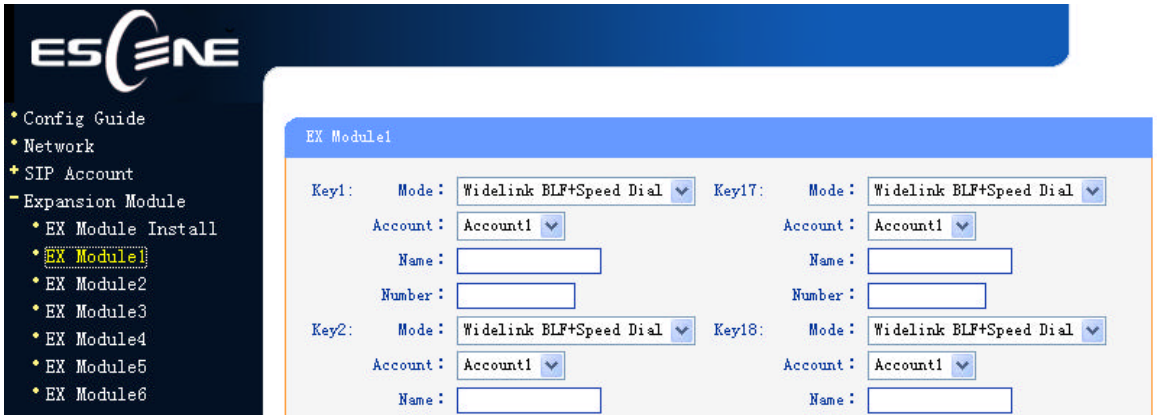
Expansion Module

Expansion module is extended Hotline function; you can believe it support more hotline by using Expansion module. You can refer to *2.Connecting Your Phone* about connecting of Expansion module.

After connecting Expansion module to a phone, you can install it as follow:



1) Click "Expansion Module", select a module that you install (the one you connect to IP phone directly is Module1, the one connect to Module1 is Module2, and so on) and click "Set".
2) After installing, you can set parameters of each Extension as follow:



Expansion module	
Key n	Each Expansion module supports 32 keys.
Mode	Five modes: --Speed Dial: Enable speed dialing in this key; --Asterisk BLF: Enable BLF in this key; --Speed Dial Prefix --DTMF --Asterisk BLA

Account	A SIP account relates to this key, another word, you will call this hotline by this SIP account.
Name	Description of this hotline.
Number	Number relates to this key.

*Regarding the settings of Expansion module, please confirm the model of your phone is ES620_E/ES620_PE.

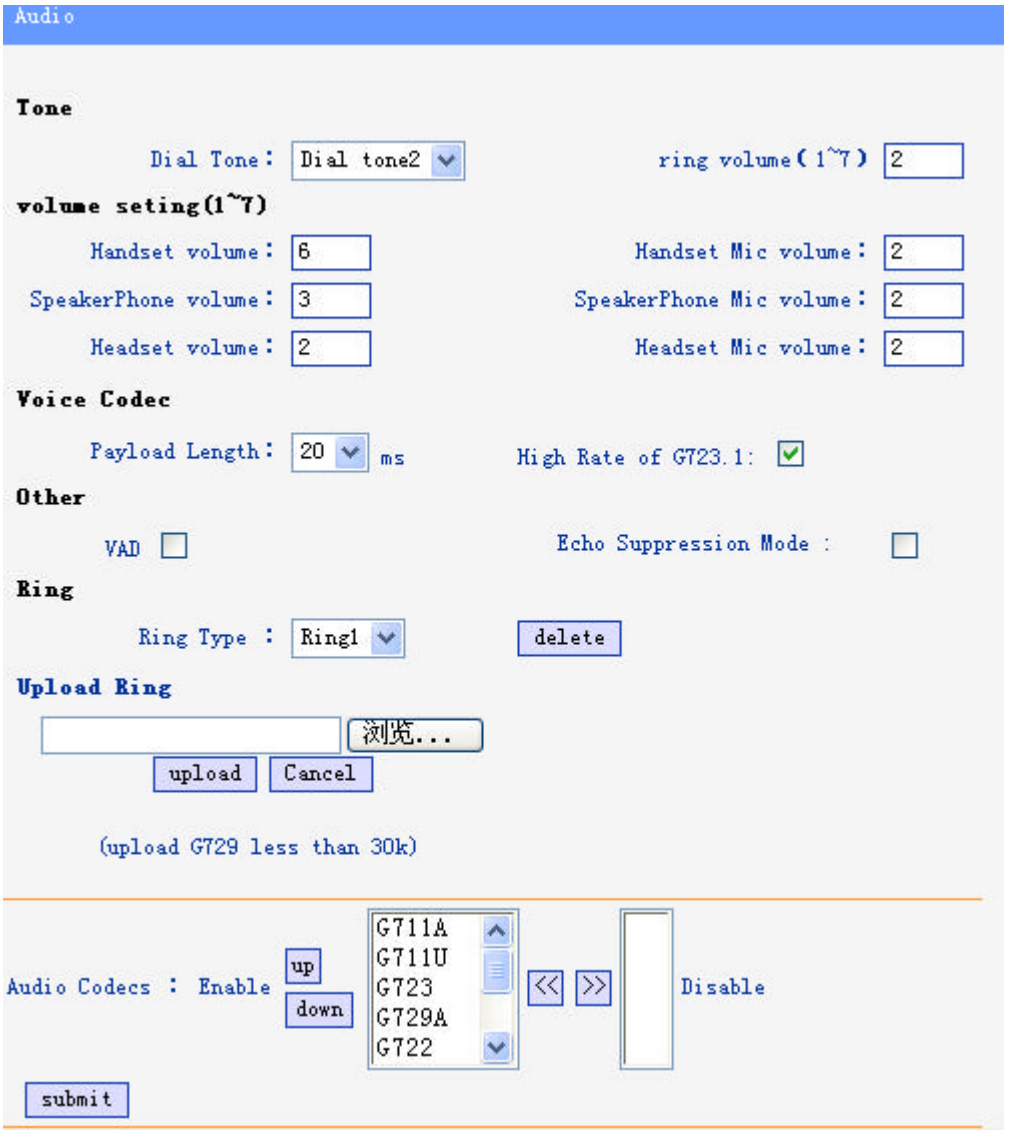
Audio

The IP phone supports the following voice codecs: G.722, G.711A, G.711U, G.723, and G.729A.

You can enable/disable the desired codecs via Web interface. Please contact your system administrator for more details about the codecs.

To enable/disable the codecs:

1) Choose Audio-> Audio Codecs



2) Use the navigation keys to highlight the desired one in the Enabled/Disable Codecs list, and press the >> / << to move to the other list.

3) Choose Submit to save the change.

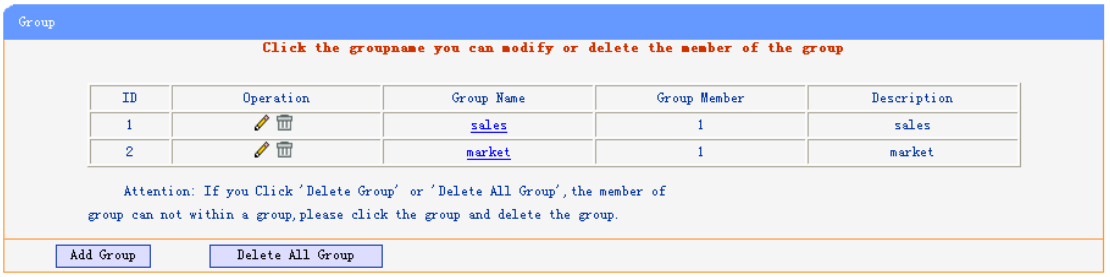
Of course, you can control the voice bulk in this choose.

PhoneBook

Group

You can add, edit and delete group in a phone book on web page of ES620.

1) Click “PhoneBook” > “Group”,



If you want to add a Group, you just ought to click ‘Add Group’.

You can edit an existed Group by click .

You can delete an existed Group by click , if you want to delete all Groups, you just ought to click ’Delete All Group’.

2) When you add a group or edit an existed group, you can set several parameters as follow:

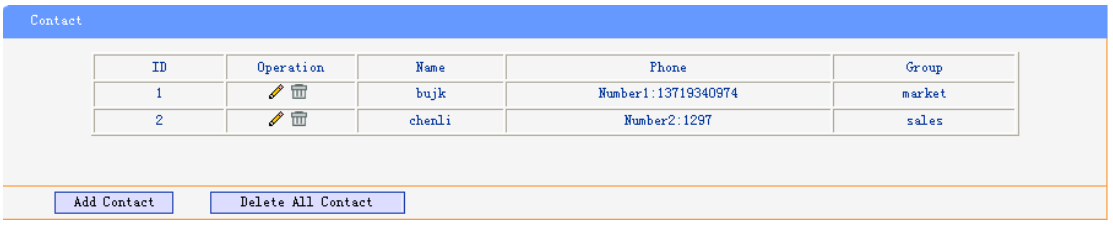
Group	
ID	Serial number of a group
Description	Description of a group
Group Name	Name of a group

Contact

You can add, edit and delete contact in a phone book on web page of ES620.

The phonebook can storage 300 contact entry.

1) Click “PhoneBook” > “Contact”,



If you want to add a Group, you just ought to click ‘Add Contact’.

You can edit an existed Contact by click .

You can delete an existed Contact by click , if you want to delete all Contacts, you just ought to click ’Delete All Contact’.

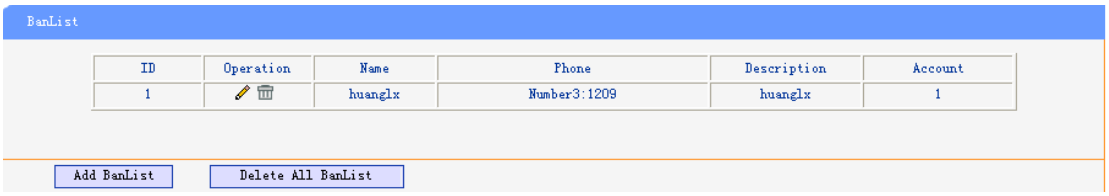
2) When you add a Contact or edit an existed Contact, you can set several parameters as follow:

Contact	
Serial Number	Serial number of a contact
First Name	The First Name of a contact
Last Name	The Last Name of a contact
Mobile Number	The Number1 phone number of a contact
Office Number	The Number2 phone number of a contact
OtherNumber	The Number3 phone number of a contact
Group	You can assign a contact to a specific group. If there isn't any group set on the phone, the group is None by default.
Account	Select a SIP account relating this contact, that is you can dial to the contact from this SIP account.

BanList

You can add, edit and delete banlist in a phone book on web page of ES620.

1) Click "PhoneBook" > "BanList",



If you want to add a BanList, you just ought to click 'Add BanList'.

You can edit an existed BanList by click .

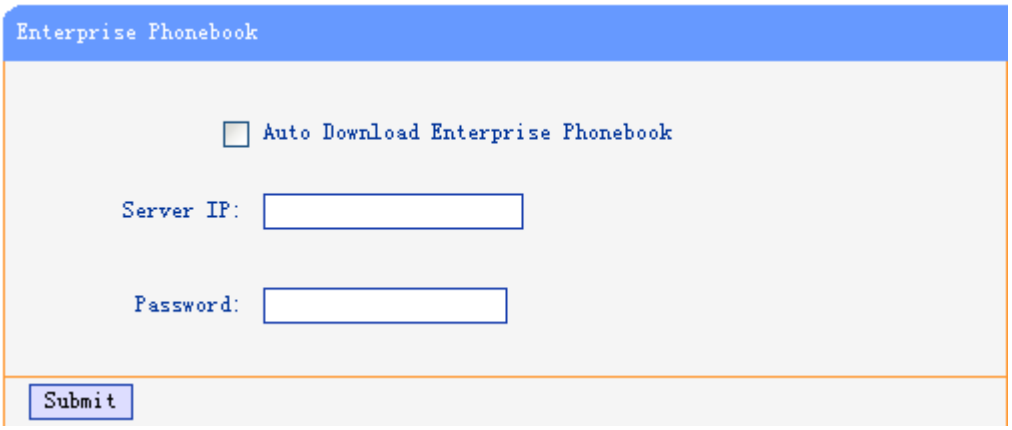
You can delete an existed BanList by click , if you want to delete all BanLists, you just ought to click 'Delete All BanList'.

2) When you add a BanList or edit an existed BanList, you can set several parameters as follow:

BanList	
Serial Number	Serial number of a BanList
Description	Description of a BanList
First Name	The First Name of a ban contact
Last Name	The Last Name of a ban contact
Mobile Number	The number1 phone number of a ban contact
Home Number	The number2 phone number of a ban contact
Office Number	The number3 phone number of a ban contact
Account	Select a SIP account relating this ban contact, that is the ban contact can't dial to this SIP account.

Enterprise Phonebook

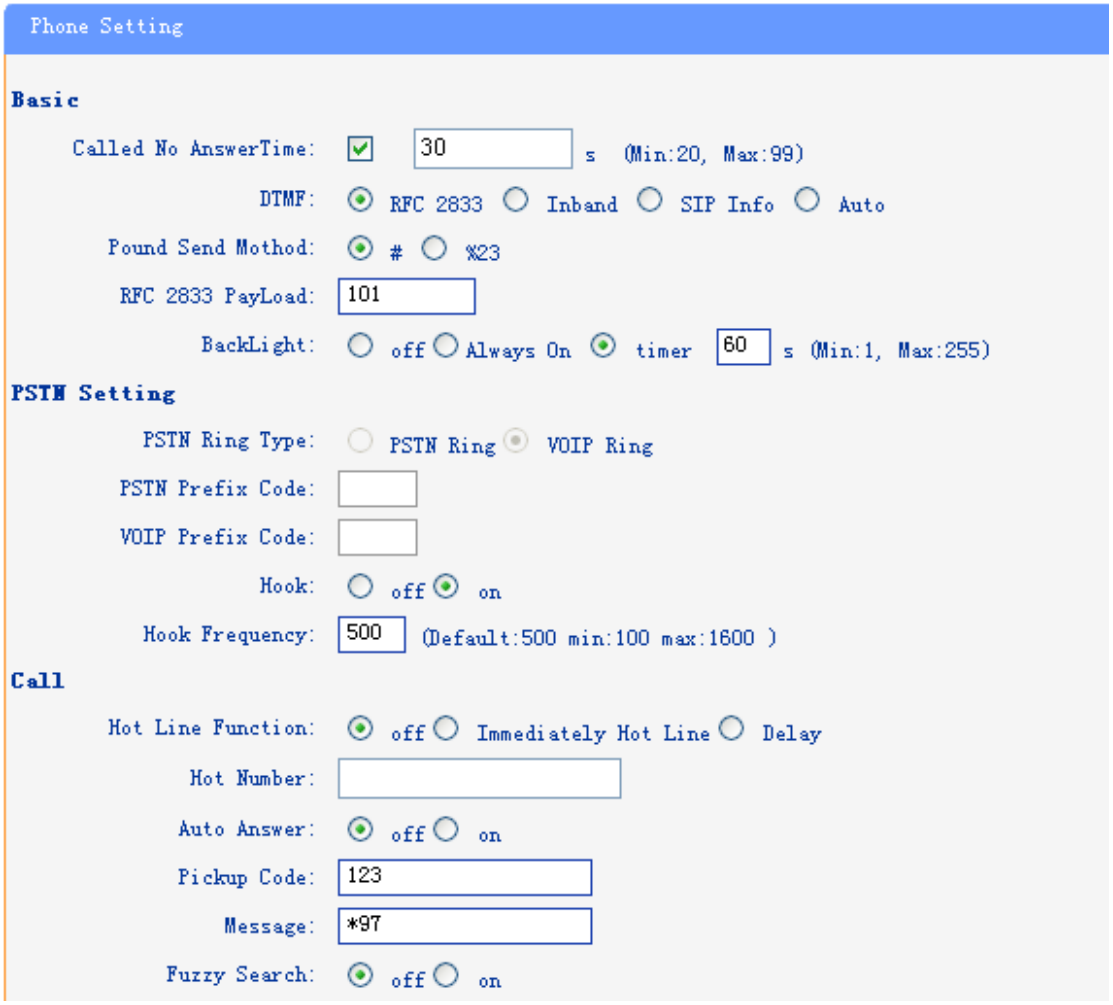
You can download Enterprise Phonebook from this web interface. But you should do second develop on the sip server to enable this function completely.
If the sip server no add some function to hold this option ,this option can be userd.



Advanced

Phone Setting

You can use phone setting to set the time,qos,port Mirroring for the phone.


Phone Setting
Basic
Called No AnswerTime: 30 s (Min:20, Max:99)
DTMF: RFC 2833 Inband SIP Info Auto
Pound Send Mothod: # %23
RFC 2833 PayLoad: 101
BackLight: off Always On timer 60 s (Min:1, Max:255)
PSTN Setting
PSTN Ring Type: PSTN Ring VOIP Ring
PSTN Prefix Code:
VOIP Prefix Code:
Hook: off on
Hook Frequency: 500 (Default:500 min:100 max:1600)
Call
Hot Line Function: off Immediately Hot Line Delay
Hot Number:
Auto Answer: off on
Pickup Code: 123
Message: *97
Fuzzy Search: off on

When used Phone Setting option,you can set several parameters as follow:

Phone Setting	
DTMF	The DTMF transmitted mode,include RFC 2833,Inband,SIP Info
BackLight	The backlight of the phone LCD
Set Time Mode	The mode of set time for phone,include SNTP/SIP Server/Manual
Daylight Saving Time	Enable/disable the DST for the phone
Time Format	You can use 24 hour time format or 12 hour time format
Time Zone-GMT	You can select different time zone for the phone
Manual Setting	This used to manual set time for the phone
QoS	The qos priority,support diff-serv and precedence
Network Packet Mirroring	When select on,then you can capture the phone's packet use notebook which connect to pc port of the phone

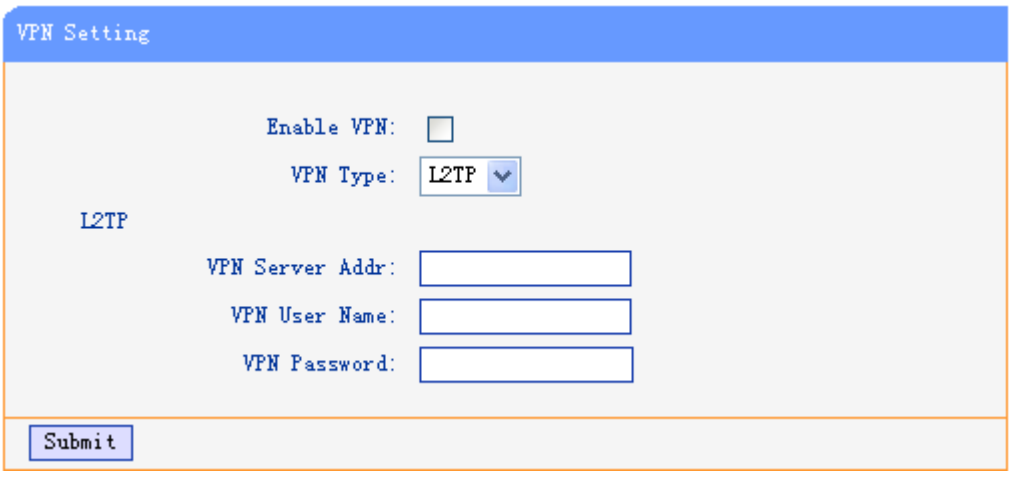
VLAN Setting

You can add the phone and PC to different VLAN used VLAN Setting option.

When used VLAN Setting option,you can set several parameters as follow:

VLAN Setting	
Enable VLAN	You can enable/disable vlan for phone and pc
VID	The vlan you want the phone or pc to join

VPN Setting



IF you need to serup a VPN Setting, you shoule fill below options.

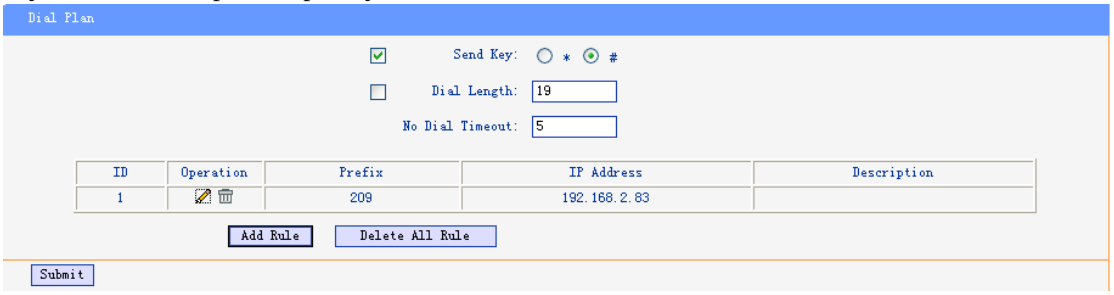
When used VPN Setting option,you can set several parameters as follow:

VLAN Setting	
Enable VPN	You can enable/disable VPN for phone and pc

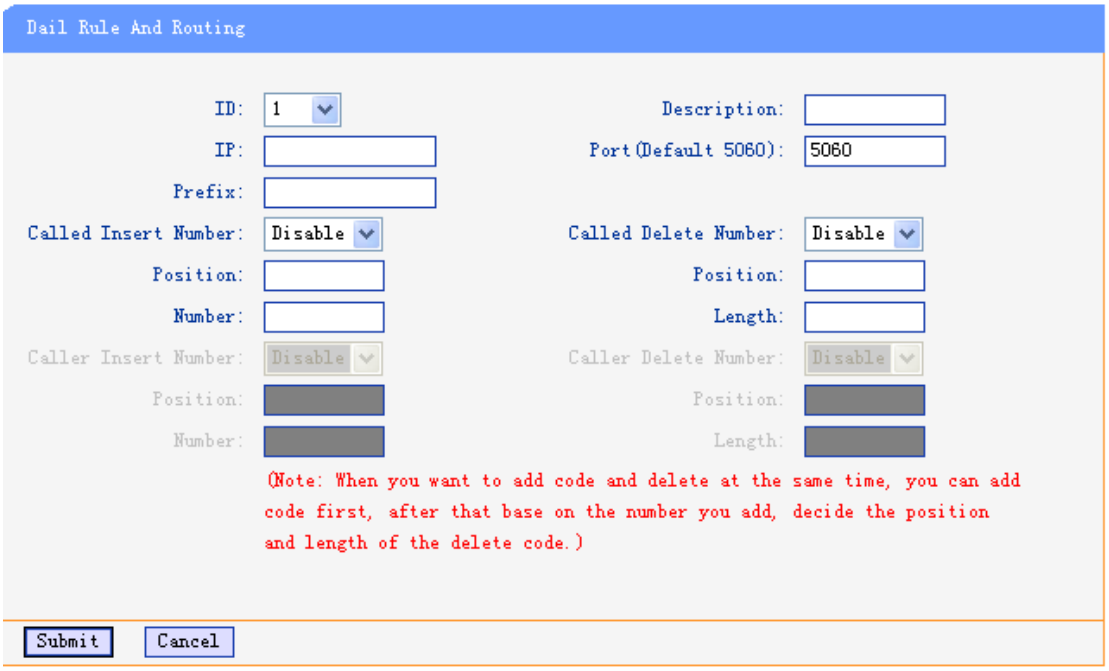
VPN Type:	There is one choose you can choice.
VPN Server Addr	VPN server'ip
VPN User Name	VPN iser's name
VPN User Name	A password be userd foe authentication

Dial Plan

If you want to setup a dial plan, you can click "Dial Plan".



Click "add rule" to entry this interface.



Dial Plan Setting	
ID	Dial Plan ID
IP	The ip of a phone which you want to call
prefix	The number which you need to press actually if you want to call the phone

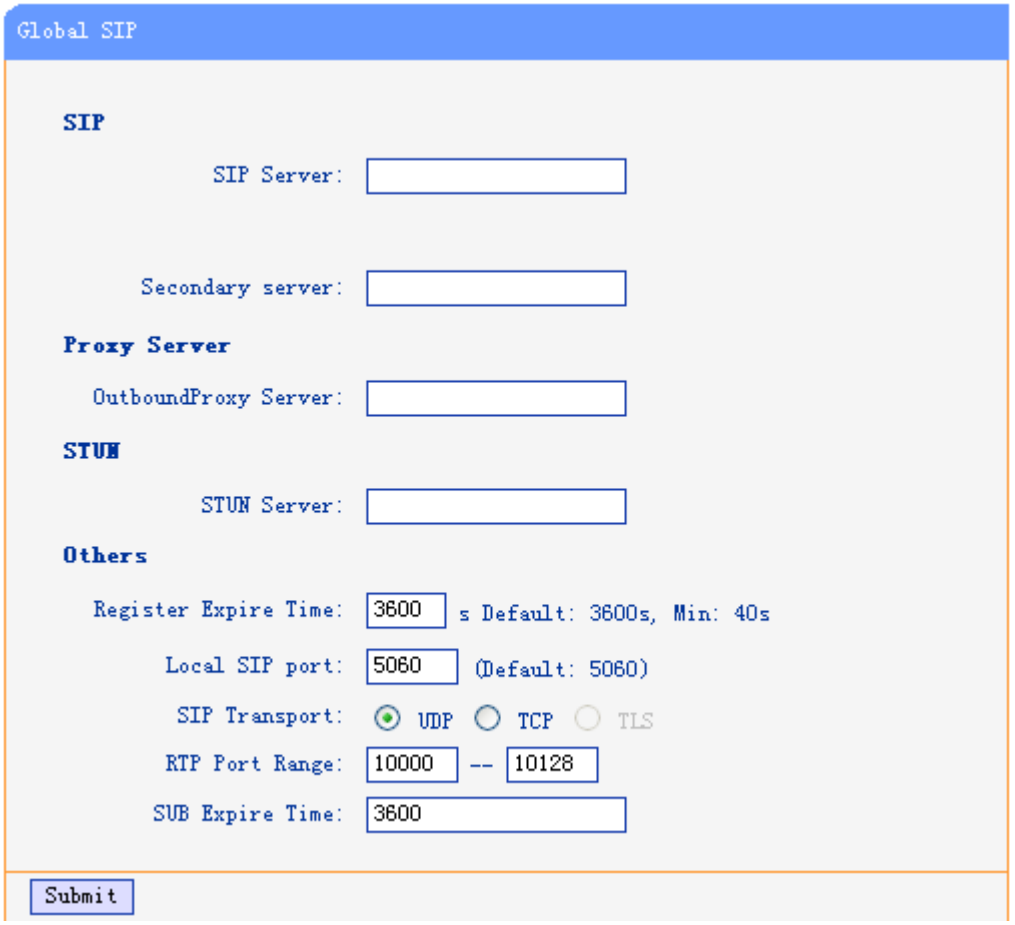
Called Number Insert	There have two option, Enable or Disable.
Position	Which position you want insert the number
Number	Waht number you want to insert
Called Number Delete	There have two option, Enable or Disable.

Tips

(Note: When you want to add code and delete at the same time, you can add code first, after that base on the number you add, decide the position and length of the delete code.)

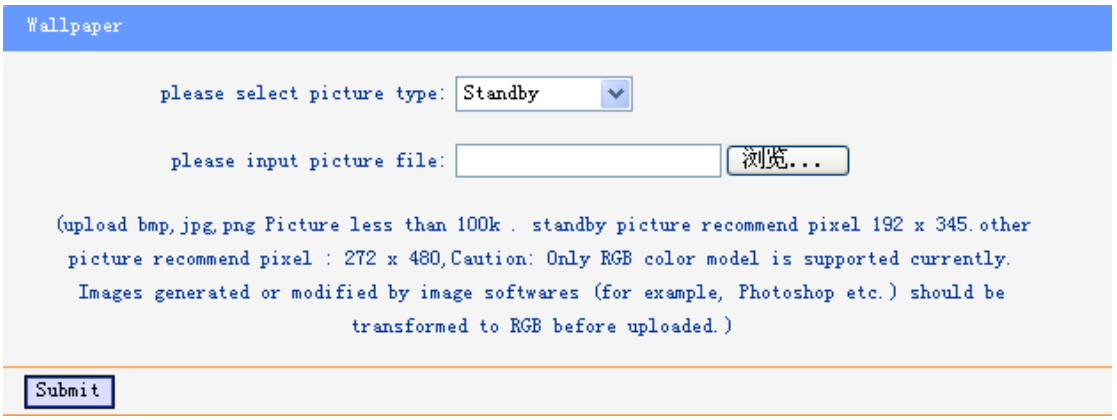
Global SIP

You also can setup the SIP server on Global SIP.



Wallpaper

You can upload picture on the phone,it can display the picture in different type.


Wallpaper
please select picture type:
Standby
please input picture file:
浏览...
(upload bmp, jpg, png Picture less than 100k . standby picture recommend pixel 192 x 345. other picture recommend pixel : 272 x 480, Caution: Only RGB color model is supported currently. Images generated or modified by image softwares (for example, Photoshop etc.) should be transformed to RGB before uploaded.)
Submit

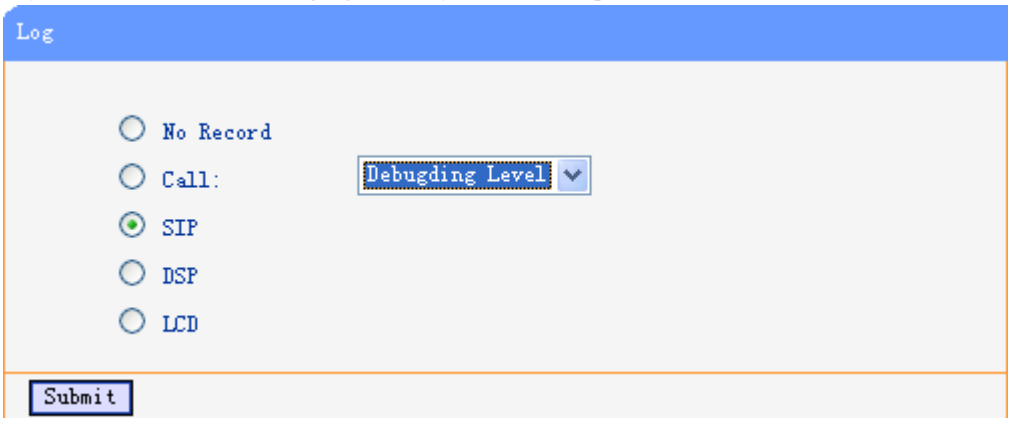
When used Wallpaper option,you can set several parameters as follow:

Wallpaper	
Please select picture type	The type of picture display on the phone LCD,it include Standby/Screensaver/Booting
Please input picture file	Select the picture you want to upload to the phone.

Phone Maintenance

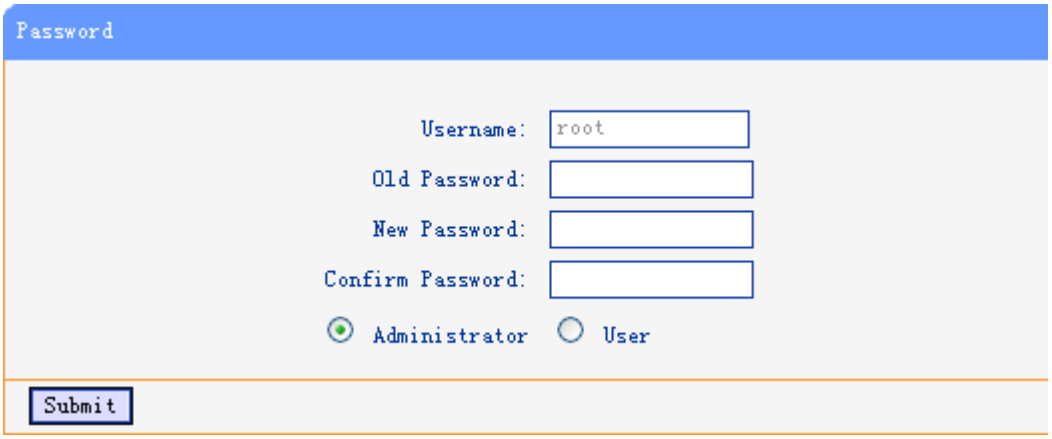
Log

If you need to catch a debuging Level, you need setup on this interface.



Password

You can change the password used to login phone GUI in Password option.

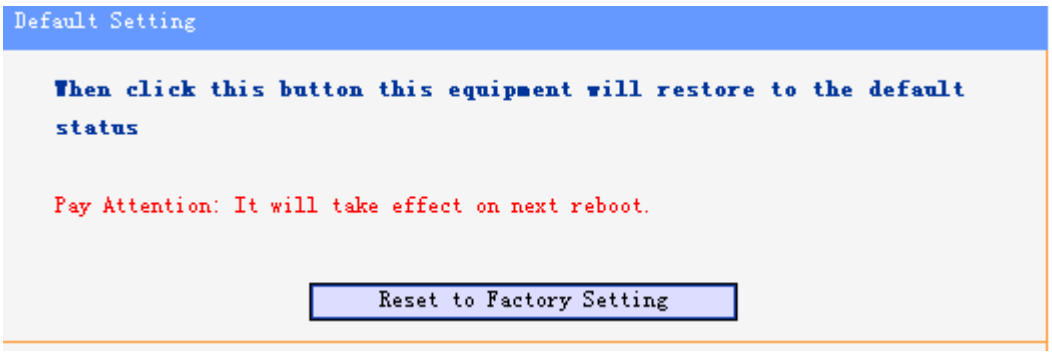


In Password option, you can set several parameters as follow:

Password	
Username	The login username of the web page
Old Password	The old password used to login of the web page
New Password	The new password used to login of the web page
Confirm Password	The new password used to login of the web page
Administrator	Login phone web page used administrator privileged
User	Login phone web page used general user privileged

Default Setting

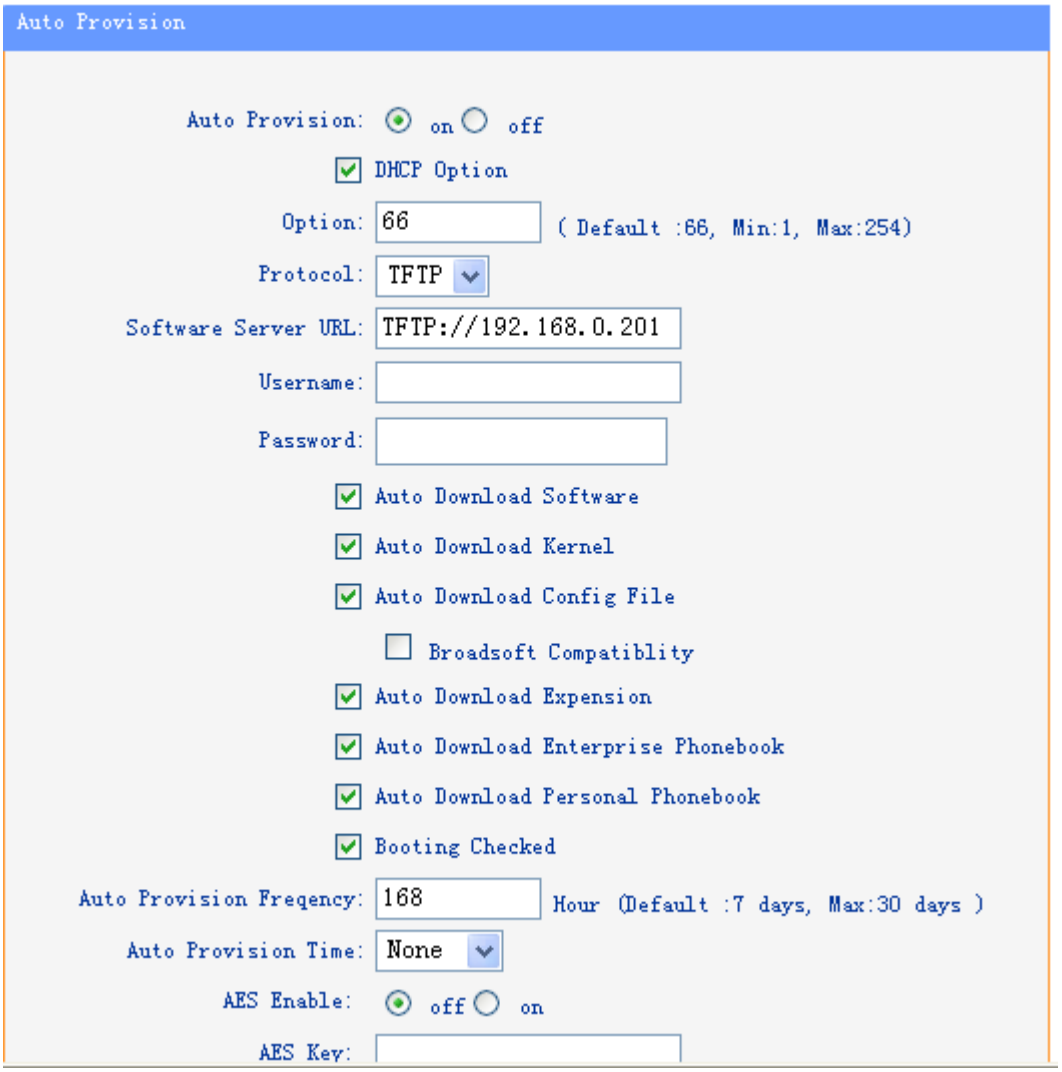
You can load the phone to the factory default setting in default setting option.


Default Setting
Then click this button this equipment will restore to the default status
Pay Attention: It will take effect on next reboot.
Reset to Factory Setting

Press the 'Reset to Factory Setting' option,the phone will load to factory default setting on next reboot.

Auto Provision

When you open the auto provision function, the phone will auto provision if the phone detect a higher sofrwre or kernel which are put on the software server. The detail information about auto provision you can see the appendix.



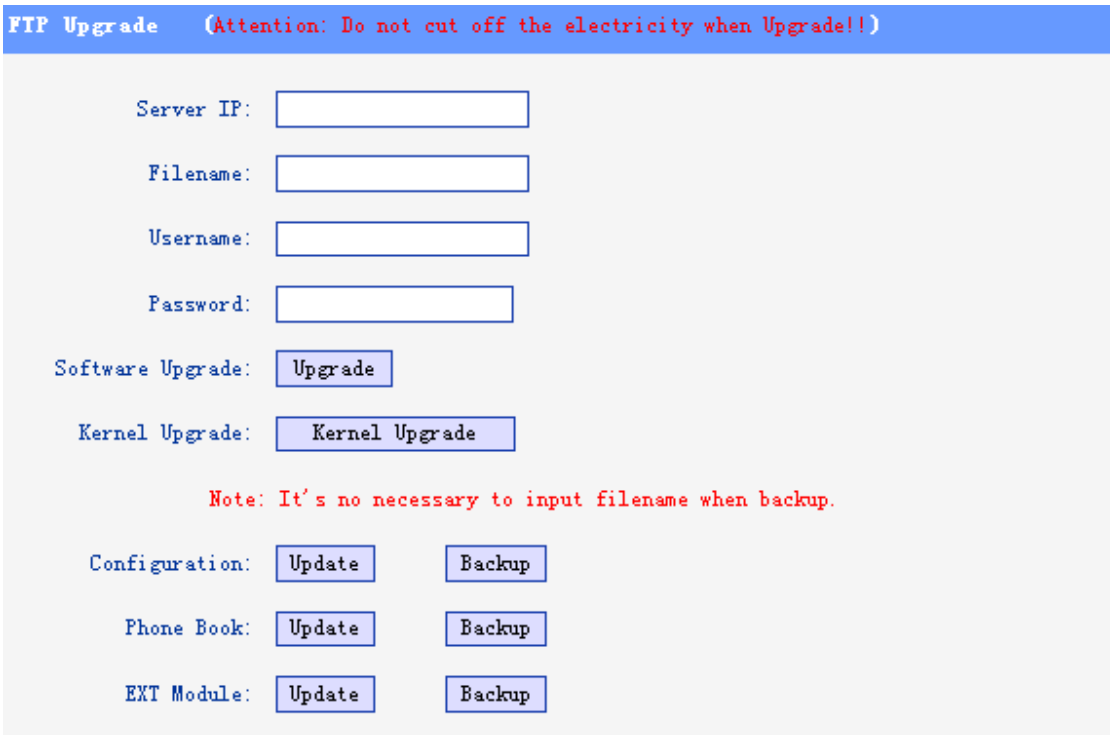
When use auto provision, you can set several parameters as follow:

Auto Provision	
Auto Provision	You can enable/disable auto provision by select on/off
Protocol	The protocol use for auto provision,it include tftp/http/ftp
Software Server URL	The server address of the auto provision
Username	The username provide by provision server
Password	The password provide by provision server
Auto Download Software	This used to auto download software from server
Auto Download Kernel	This used to auto download kernel from server

Auto Download Config File	This used to auto download config file from server
Broadsoft Compatiblity	This used to compatible the broadsoft format's config file
Auto Download Expension	This used to auto download expansion's config from server
Auto Download Enterprise Phonebook	This used to auto download enterprise phone from server
Auto Download Personal Phonebook	This used to auto download personal phonebook from server
Booting Checked	This used to checked the auto provision when phone booting
Auto Provision Freqency	This used to set the time interval for auto provision
Auto Provision Time	This used to the specific time for auto provision
AES Enable	You can enable/disable AES encrypt for auto provision
AES Key	The key of the AES
Auto Provision Now	This used to do auto provision immediately

FTP Upgrade

You can upgrade the software,kernel and configure file for the phone use ftp.

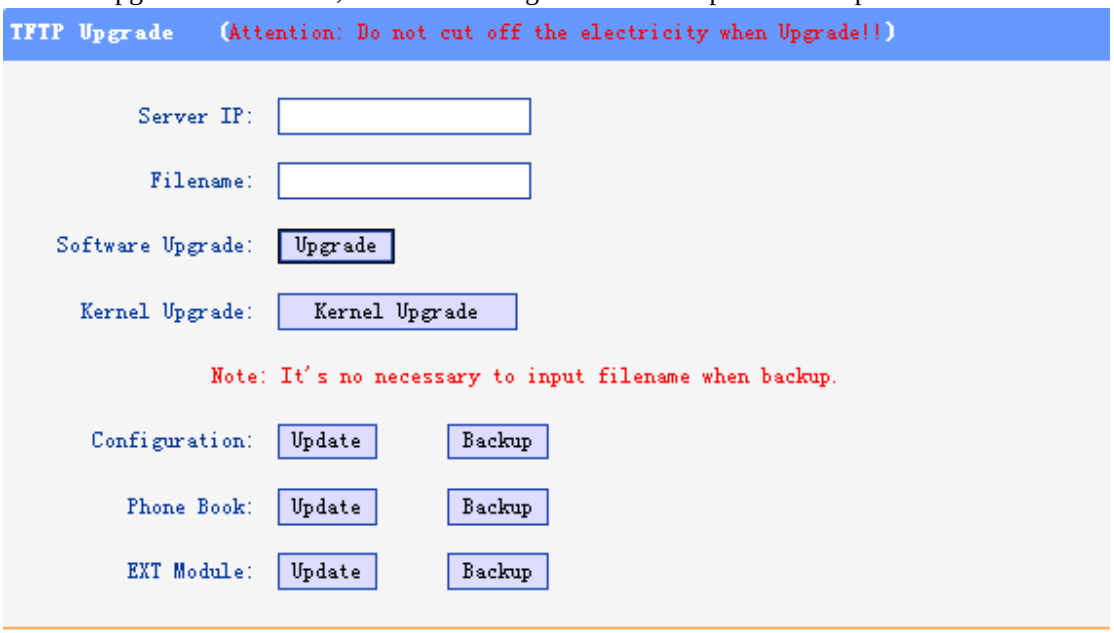


When use ftp upgrade, you can set several parameters as follow:

FTP Upgrade	
Server IP	The ip address of the ftp server
Filename	The name of the file want to download from ftp server
Username	The username provide by ftp server
Password	The password provide by ftp server
Software Upgrade	Used to upgrade the software of the phone
Kernel Upgrade	Used to upgrade the kernel of the phone
Configuration	You can used update/backup to update/backup the configure file of the phone
Phone Book	You can used update/backup to update/backup the phonebook of the phone
EXT Module	You can used update/backup to update/backup the expansion of the phone

TFTP Up grade

You can upgrade the software,kernel and configure file for the phone use tftp.



When use tftp upgrade, you can set several parameters as follow:

TFTP Upgrade	
Server IP	The ip address of the tftp server
Filename	The name of the file want to download from ftp server
Software Upgrade	Used to upgrade the software of the phone
Kernel Upgrade	Used to upgrade the kernel of the phone

Configuration	You can used update/backup to update/backup the configure file of the phone
Phone Book	You can used update/backup to update/backup the phonebook of the phone
EXT Module	You can used update/backup to update/backup the expansion of the phone

HTTP Upgrade

You can upgrade the software,kernel and configure file for the phone use http.

When use http upgrade, you can set several parameters as follow:

HTTP Upgrade	
Select a File	Browse the software/kernel/config file you want to upgrade from http
Software Upgrade	Used to upgrade the software of the phone
Kernel Upgrade	Used to upgrade the kernel of the phone
Configuration	You can used upload/download to upload/download the configure file of the phone
Phone Book	You can used upload/download to upload/download the phonebook of the phone
EXT Module	You can used update/backup to update/backup the expansion of the phone

Reboot

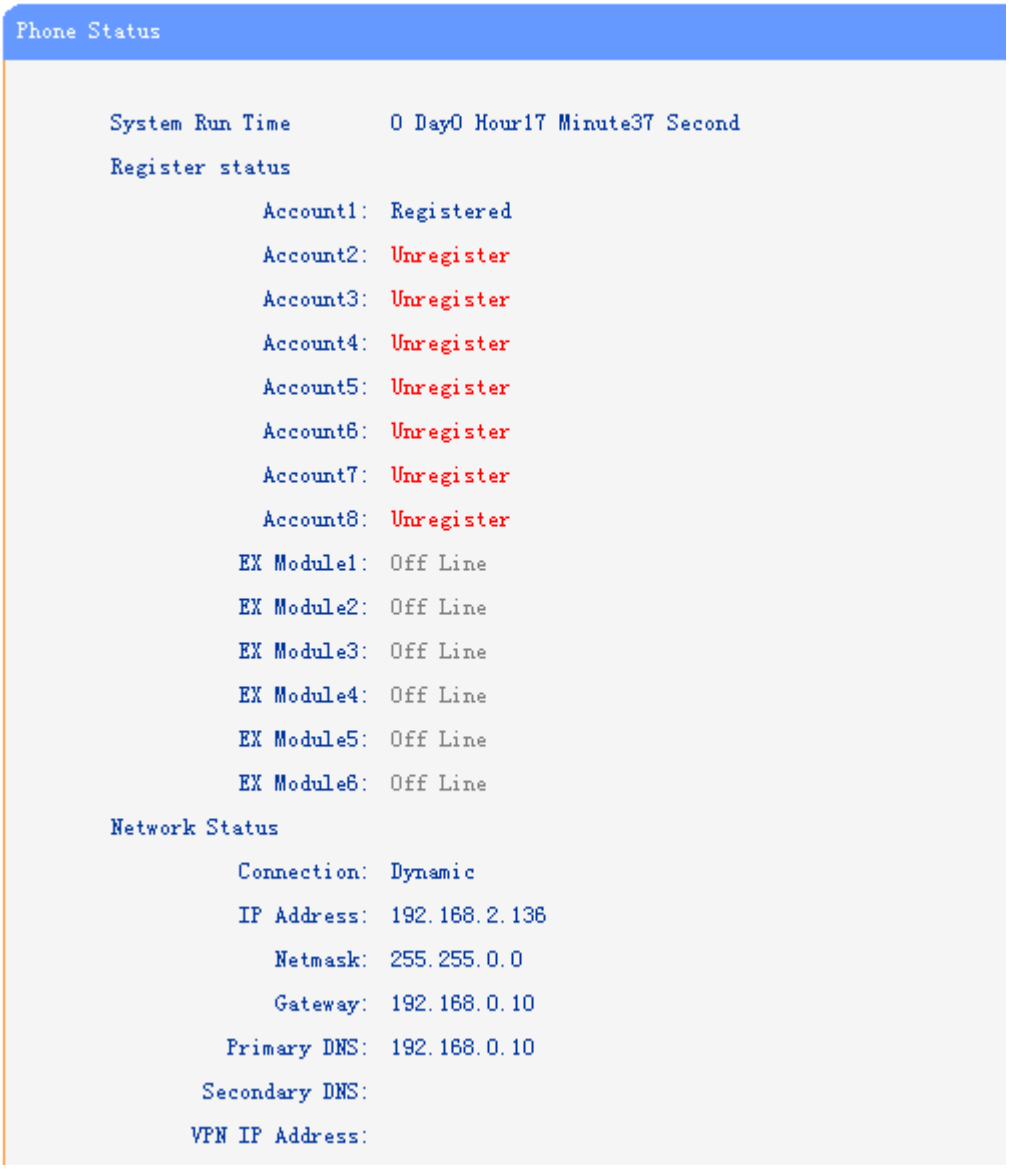
You can use reboot option to reboot the phone.

Reboot
Attention: When click this button this equipment will be reboot, web service will be interred, please connect again.
Reboot

When you press ‘Reboot’,the phone will reboot.

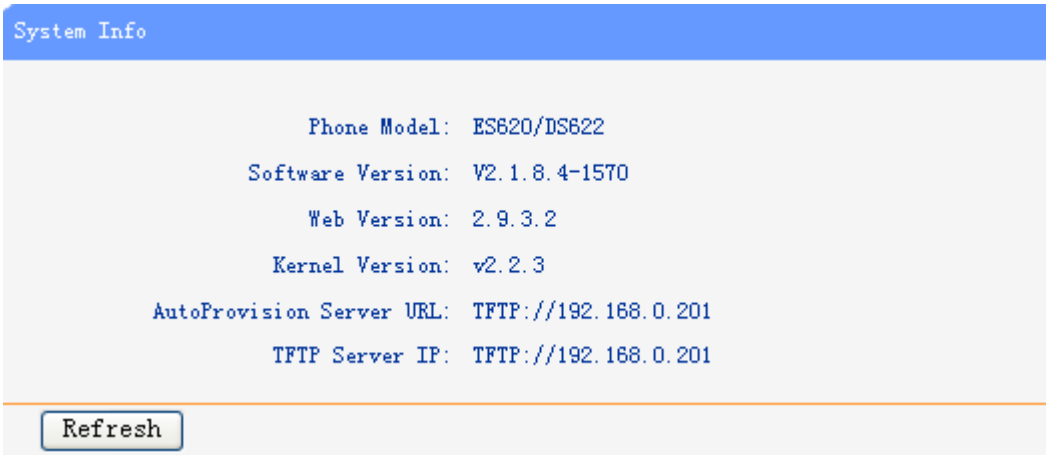
Phone Status

You can see the currently status of the phone when use Phone Status option.



System Info

You can see the system information when used System Info option.



Attention:
On this interface ,you can see the software and kernel which we used for test and this user_manual is written base on this software and kernel.
This software version is V2.1.8.4-1570
This kernel version is v2.2.3

About

You can see the phone model when used About option.

Appendix：

Auto Provision

Pre-configuration on TFTP/HTTP/HTTPS/FTP Server

When the software or kernel auto-provision is enabled and want to run, IP Phone will check the software and kernel version at first, so we need make some pre-configuration on the provisioning server.

Auto Provision for Software:

1. Create a notepad file named"**F000X00.cfg**"(the "X" is decided by the model of the IP phone you are using, for example, if the model is ES620, the file name is "F000600.cfg");

   **Named rule of the file:*

   F00600.cfg: for ES620, ES610 and ES620;
   F00400.cfg: for ES410 and DS412;
   F00300.cfg: for ES310 and DS312;
   F00200.cfg: for ES210 and DS212.

2. Open the notepad file "F000X00.cfg" and write the new software name in it, for example,

   S_ES6xx_version2.0.4.6: for ES620, ES610 and DS622;
   S_ES410_version2.0.4.6: for ES410 and DS412;
   S_ES310_version2.0.4.6: for ES310 and DS312;
   S_ES210_version2.0.4.6: for ES210 and DS212

   Write down the new version you want to upgrade and save it on your provisioning server.

   **Please note that if the version is not older than (and same as) the one on your phone, auto-provision of your software would be not available.*

3. After it, upload the new software to the TFTP/HTTP/HTTPS/FTP provisioning server and complete the pre-configuration steps.

Auto Provision for Kernel:

1. Create a notepad file named"**K000X00.cfg**"(the "X" is decided by the model of the IP phone you are using, for example, if the model is ES620, the file name is "K000600.cfg");

   **Named rule of the file:*

   K00600.cfg: for ES620, ES610 and DS622;
   K00400.cfg: for ES410 and DS412;
   K00300.cfg: for ES310 and DS312;
   K00200.cfg: for ES210 and DS212.

2. Open the notepad file "K000X00.cfg" and write the new kernel name in it, for example,

   K_uImage_600.bin_version2.1.6: for ES620, ES610 and DS622;
   K_uImage_400.bin_version2.1.6: for ES410 and DS412;

K_uImage_300.bin_version2.1.6: for ES310 and DS312;
K_uImage_200.bin_version2.1.6: for ES210 and DS212

Write the new version you want to upgrade and save it on your provisioning server.

**Please note that if the version is not older than (and same as) the one on your phone, auto-provision of your kernel would be not available.*

3. After it, upload the new kernel to the TFTP/HTTP/HTTPS/FTP provisioning server and complete the pre-configuration steps.

Configuration files on TFTP/HTTP/HTTPS/FTP Server

- **Name of configuration file:**

The configuration file on the provisioning server is named as the MAC address of IP phone itself. Escene's IP phones support two different configuration files for auto-provision:

1. Normal Configuration file:

Normal Configuration file is the configuration file of your Escene IP phone. You can download it from your phone (You can see the following chapter to see how to download a configuration file from Escene IP hone) and modify by yourself. If the IP phone's MAC address is 00:11:22:33:44:55, the normal configuration file of it should be ***001122334455.xml***.

2. Broadsoft Configuration files:

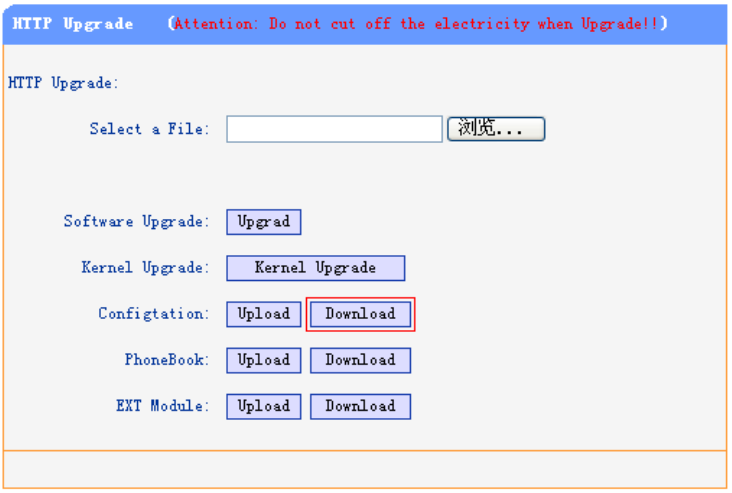
Broadsoft Configuration files support the format of Broadsoft IP-PBX. However, you can use them for provisioning. There are two files should be set on your provisioning server, they are also named by the MAC address of your phone

1) ***001122334455.cfg***: a configuration file for system settings, for example, network, audio and so on.
2) ***001122334455.txt***: a configuration file for SIP accounts.

- **Download a configuration file from your phone:**

You can download a configuration file from your phone by HTTP as follow:

1. Open the web page of your IP phone, click "Phone Maintenance">"HTTP Upgrade";
2. Then click "Download" of Configuration:



3. If you want to use this file to auto-provision, you just need to modify it by yourself and rename it to the MAC address of your IP Phone with .xml suffix.

Extern.xml file on TFTP/HTTP/HTTPS/FTP Server

The Extern.xml includes the settings of programmable buttons on the phone and all Expansion Modules. All the phones can download the settings from a same file and they will have the same settings (for example, Speed-dial, BLF and so on).

You can't rename the file on the provisioning server. The file name is fixed to* ***Account1_Extern.xml**.(Account1 is the first account you register)*

Phonebook on TFTP/HTTP/HTTPS/FTP Server

Escene IP phone supports Enterprise Phonebook and Personal Phonebook.

- **Enterprise Phonebook:**

Enterprise Phonebook is used for all staffs in your office. All phones will download a common phonebook for all staffs. The file's name must be ***Enterprise_Phonebook.xml*** on your provisioning server and you can not rename it.

- **Personal Phonebook:**

Personal Phonebook is individual for each IP phone. The file on your provisioning server is named by the first account of your IP phone. If the IP phone's first account is 1287, the Personal Phonebook of this phone is ***1287_Phonebook.xml***.

Automatic Provisioning using DHCP Option 66

The following steps will descript auto-provision by TFTP. You also can use HTTP and FTP for auto-provision with our phones.

DHCP Server: (Microsoft Windows 2003 server)

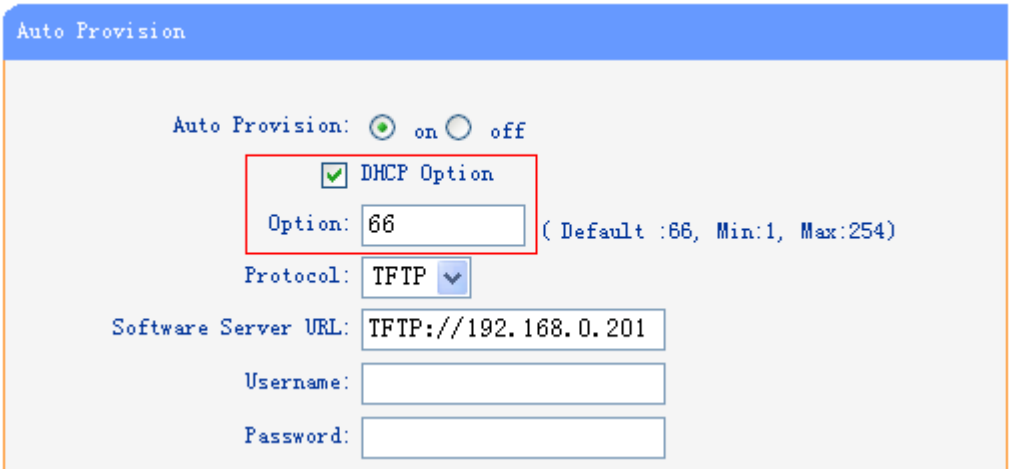
1. Start up the "DHCP Management Console";
2. Expand the DHCP scope which will contain the phones
3. Right-click on the "Scope Options" node
4. Select "Configure Options"
5. In the "General" tab, scroll down the list of options and identify the option labeled "066 Boot Server Host Name"
6. Enable the "066 Boot Server Host Name" and enter the string value according to the examples discussed previously

string value: 192.168.0.201(TFTP Server IP Address);

7. Click the "OK" button

IP Phone:

1. Input the IP Phone's IP Address in browser;
2. Enter user and password with "root" then open the web page;
3. Click "Phone Maintenance" and select "Auto Provision";
4. Select like as follows:

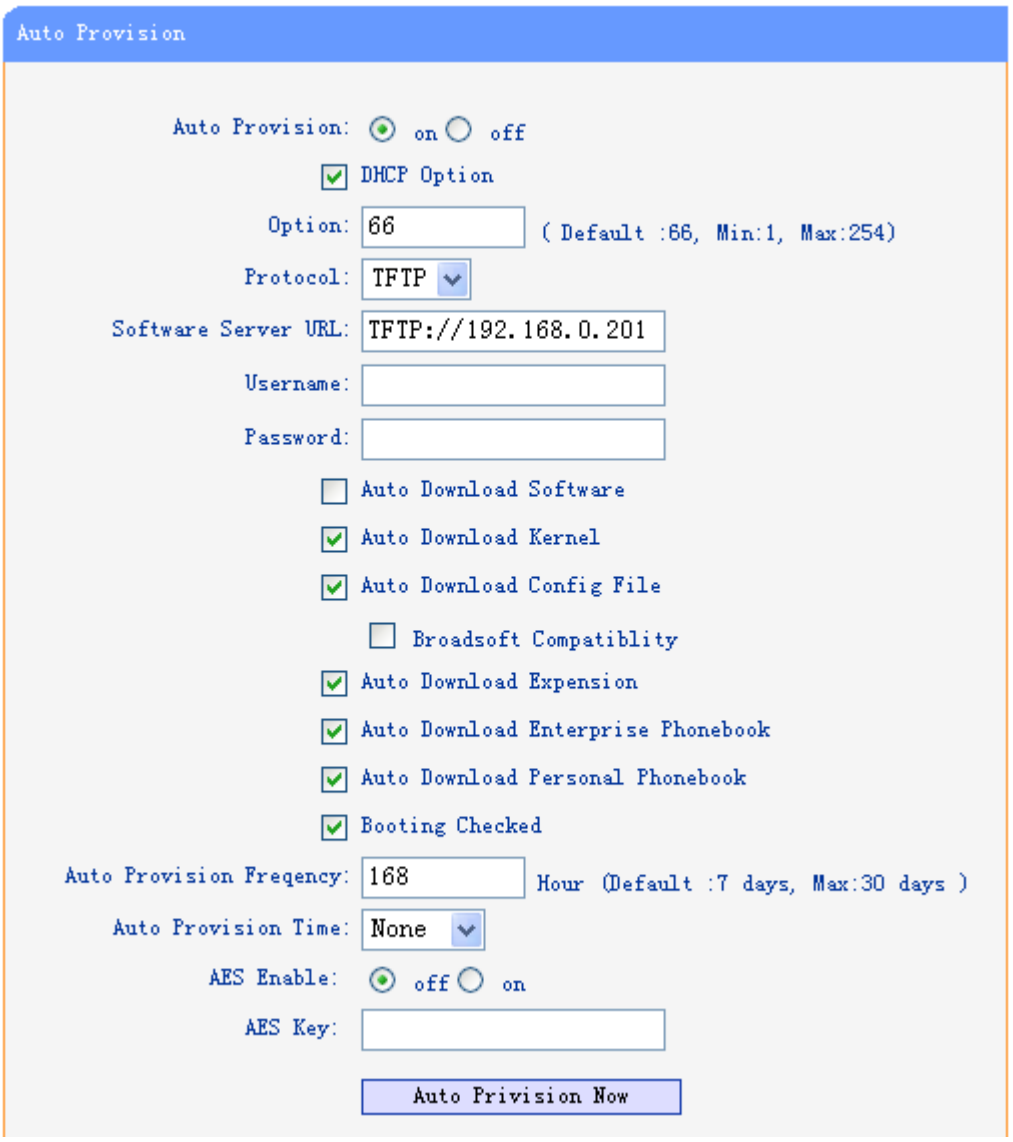


5. Click "Submit" to save it.

Auto-Provision via fixable TFTP/HTTP/HTTPS/FTP Server

IP Phone:

1. Input the IP Phone's IP Address in browser;
2. Enter user and password with "root" then open the web page;
3. Click "Phone Maintenance" and select "Auto Provision";
4. select like as follows:



It supports three protocols in Auto-Provision:TFTP,HTTP and FTP.
The format with provisioning server URL is:

TFTP:

TFTP://192.168.0.201(192.168.0.201 is the default Server IP address)

HTTP:

HTTP://192.168.0.201

HTTPS:

HTTPS://192.168.0.201

FTP:

FTP://192.168.0.201

Username: the user to login FTP/HTTP/HTTPS server

Password: the password of the user using to login FTP/HTTP/HTTPS server

***Username and password are available in FTP/HTTP/HTTPS only (unavailable in TFTP).*

Auto Download Software:
Download software from server and upgrade it automatically.
Auto Download Kernel:
Download kernel from server and upgrade it automatically.

Auto Download Config File:
Download configuration file from server and update it automatically.
BroadsoftCompatibility:
If you select this function, you need to put two configuration files (with Broadsoft format) on the provisioning server. Otherwise, you can download the configuration file from your phone via HTTP (regarding the steps, you can refer to *" Download a configuration file from your phone"* in this document.), modify it and upload it to the server for auto-provision.
Auto Download Expansion:
Download configuration file of the Programmable buttons on your phone or Expansion Modules automatically.
Auto Download Enterprise Phonebook:
Download Enterprise Phonebook from server and update it automatically.
Auto Download Personal Phonebook:
Download Personal Phonebook from server and update it automatically.
Booting Checked:
Check all items you had selected and upgrade/update them when the phone boot
Auto Provision Frequency:
The auto provision Frequency which you want.
Auto Provision Time:
The time you want to execute auto-provision.

Examples of Auto Provision Frequency and Time

1) When you set the **Auto Provision Frequency** and disable **Auto Provision Time** (set to None), the Auto Provision function will work after the **AutoProvision Frequency**;
2) When you set both **Auto Provision Frequency** and **Auto Provision Time**, for example:
   You set the **AutoProvision Frequency** to 24 hours, and the **Auto Provision Time** to 2:00 at 8:00 today (1, Jan), it will pass 24 hours at first and work at the nearest 2:00, it means that the Auto Provision function will work at 2:00 on the day after tomorrow (3, Jan).
   Therefore, if you want this function work at 23:00 tonight and it is 8:00 now, you need to set the **Auto Provision Frequency** to 0 hours and the **Auto Provision Time** to 23:00.

AES Encryption:
AES encryption is used for all the setting files of your phone (include configuration file, Expansion file, Enterprise/Personal Phonebook etc. You just need to enable the

AES Encryption function and input the AES Key matching the one on your server on.

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